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Application for Patent

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אני, (שם המבקש, מענו -- ולגבי גוף מאוגד-- מקום התאגדותו)
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ששמה הוא Being the Inventor
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TWO-WAY GROUP COMMUNICATION SYSTEMS (באנגלית) (English)

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מבקשת פטנט from Application	*לבקשה/לפטנט to Patent/Appl.	מספר/סימן Number/Mark	תאריך Date	מדינת האיגוד Convention Country
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Signature of Applicant

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Sanford T. Colb & Co.
C:27662

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מערכות קשר דו-כיווני קבוצתי

TWO-WAY GROUP COMMUNICATION SYSTEMS

DORON SHALEV

דורון שלו

C:27662

Two-Way Group Communication Systems

Field Of The Invention

The present invention relates to two-way group communication systems and more particularly, but not exclusively, to a method and apparatus for providing wireless communication between a plurality of communication devices forming a group, such that each device hears each other device in the group.

Background Of The Invention

Wireless Intercom is a form of radio transmission which allows a group of users, or stations, to communicate with one another. Communication is many-to-many so that all the users can hear anyone who is transmitting at any given time. For regulatory reasons many wireless intercom systems use spread spectrum implementations.

General considerations of radio communications, such as bandwidth usage, apply to wireless intercom. Such general considerations, and in particular general considerations as they apply to spread spectrum, may be found in The ARRL Spread Spectrum Sourcebook, First Edition 1991 ISBN 0-87259-317-7, and in Spread Spectrum Systems with Commercial Applications - R. Dixon 3rd Edition 1994, ISBN 0-471-59342-7. A particular feature of an intercom system is that more than one group member is able to transmit at the same time. In addition each group member must be able to receive each other group member.

There are several types of intercom system currently in use. US 4,470,141 discloses a wireless intercom system involving a single base station and one or more mobile stations. The base station assigns a channel to each mobile station so that each mobile station is able to transmit to the base station without fear of interference from the other mobile stations. The base station then adds the signals from each mobile station to produce a composite signal that is transmitted to each mobile station. Thus each user is enabled to hear each other user. The disadvantages of this system are that it requires multiple channels for operation and that it requires a large and relatively immobile base station. If the base station fails then the entire system fails. There is no possibility of using any of the mobile stations as an impromptu base station since the mobile stations cannot manage the number of channels necessary. In current systems, frequency hopping may be employed but this

does not alter the total number of channels needed at any given time. Such a solution is known as a star connection, since each individual mobile station communicates only with the central base station.

In summary, in a star system the total number of channels needed is one duplex channel per mobile user. The base station must be able to operate all of the channels being used and as such is large and immobile. In the event of its failure the mobile stations are useless. In addition current systems do not provide any means of judging how near or how far respective users are.

US 5,274,666 discloses an intercom system involving a group of mobile stations arranged in a loop. Each mobile station receives a signal from a preselected one of the other mobile stations, adds any signal of its own and then passes the signal to another preselected one of the mobile stations. The group of stations thus constitute a logical loop and there is no need for a base station. However each station transmits on a different frequency and if one of the stations should fail then the entire loop has to be reconfigured. In particular there is a possible condition known with this system in which control signals are passed on by a particular mobile station but the voice signal is not. In this case the failure is not detected and the system does not reconfigure. Each mobile station will then receive all signals added between the faulty station and itself in the flow direction of the loop but no others. It may thus take a while for users to become aware that there is a problem. Again the system may be adapted to use frequency hopping, as shown in US 5,274,666. Such a solution is known as a ring connection.

There are numerous variations and sub-combinations of the two basic intercom systems mentioned above. However the above summarizes the basic principles involved in the art.

Summary of the Invention

According to a first aspect of the present invention there is provided a method of simultaneously recovering a plurality of double sideband radio signals that are not mutually synchronized. The method comprises the steps of receiving the signals and demodulating the signals using the output of a phase shifting device.

Preferably the output of the phase shifting device is predetermined to have an average phase which cancels the effect of a cosine term in the demodulated signal to render the plurality of

received signals audible. In an embodiment this is achieved in that the phase shifting device produces an output phase which is proportional to a phase control signal and wherein the phase shifting device receives an input which has a constant phase. The phase control signal may be from a pseudo-random generator.

Preferably the radio signals are spread spectrum radio signals.

Any of the above mentioned radio signals may be AM (Amplitude Modulated) signals.

According to a second aspect of the invention there is provided a method of simultaneously recovering a plurality of single sideband radio signals that are not mutually synchronized, comprising the steps of receiving the signals and demodulating the signals.

Preferably the radio signals are spread spectrum radio signals.

According to a third aspect of the invention there is provided a method of sending an electronic signal from a first point, or slave to a second point or master remote from the first point and providing an indication to the first point regarding receipt of the signal at the second point, comprising the steps of relaying the electronic signal back from the second point to the first point and monitoring the level of the relayed signal at the first point.

Preferably the first point has a sound input site and a sound output site. The sound is converted into the electronic signal after passing through the sound input site. Thus the relayed signal may be monitored by converting the relayed signal into sound at the sound output site and listening to the sound. If the sound is faint then it is clear to the user of the communication device that he is far away from the second point or that there is some sort of transmission problem and he is able to take any necessary action. Preferably the electronic signal is passed to the sound output site only after being relayed from the second point, and no direct connection is provided within the communication device itself between the sound input and the sound output. This prevents any problems to do with acoustic feedback and moreover allows a judgment to be made as to how well the signal has been received.

Preferably, the level of the sound output from the sound output site is allowed to vary with the level of the relayed signal so as to reflect the average level of the signal. Again this aids the user in making the above-mentioned judgments. It is also desirable that the second point have a sound output site to which the electronic signal is directed to for conversion into sound. Again, the level of the sound at the sound output site of the second point, resulting from the conversion, may be allowed to vary with the level of the electronic signal to reflect the average level of the signal.

Again, preferably the electronic signal is passed from the second point to at least one other point, wherein the one other point has a sound output site, wherein sound is output at the sound output site in response to the electronic signal and wherein the level of the sound output at the sound output site is allowed to vary with the level of the electronic signal to reflect the average level of the signal. Thus users of the other slave devices are able to make judgments about the nearness of other users.

Preferably the sound output by the first point is permitted to vary in strength according to the strength of a respective received signal. Thus the user is able to make judgments about receipt of his own signal by the network.

According to a fourth aspect of the present invention there is provided an apparatus for simultaneously recovering a plurality of double sideband radio signals that are not mutually synchronized. The apparatus comprising a demodulation device and a phase shifter device. Preferably the phase shifter has two input connections and produces a phase shift according to a level of a signal received at the second input connection. The first input connection would thus be connected to a device that produces a signal which has a constant phase. Preferably the first input of the phase shifter is connected to the output of an oscillator. The second input connection would thus be connected to a device adapted to produce a signal having a constant average level. Preferably the second input of the phase shifter is connected to the output of a pseudo random generator.

According to a fifth aspect of the present invention there is provided a method of providing wireless communication between a plurality of communication devices forming a group, such that each device receives transmissions from each other device in the group, comprising the steps of selecting one of the devices as a master, hence designating all other devices as slaves, transmitting

from the master to each slave using a preselected channel, synchronizing each slave to the master, transmitting from each slave to the master using the preselected channel, combining the signals from each slave at the master, and transmitting the combined signals as a single signal from the master to each slave to be received at each slave.

Preferably transmission between the master and the slaves is carried out using time division duplex and spread spectrum.

In an embodiment the signals from the slaves are not mutually synchronized. Hence, upon receipt at the master, they are demodulated using a pseudo-randomly phase shifted LO (Local Oscillator).

There is not much difference between the circuitry required for operation as a master and for operation as a slave. Hence it is easy to ensure that failure of the master leads to selection of one of the slaves to serve as master.

Each slave preferably has a sound input site and a sound output site, wherein sound is converted into an electronic signal after passing through the sound input site, wherein the electronic signal is transmitted to the master, wherein the signal received from the master includes the electronic signal sent to the master, and wherein the received signal is converted into sound at the sound output site.

In a preferred embodiment a tone is added to signals transmitted by the master and the tone is used by at least one of the slaves to correlate with the master. The signals transmitted with the tone may be shaped using a Gaussian curve.

A facility may also be provided for shifting a base frequency used for transmission upon detection by the master of other groups operating in the vicinity.

A communication device for use in any of the above methods may comprise circuitry for operation as a slave and additional circuitry for operation as a master, together with a switching circuit for switching between master and slave operation. The switching circuit may be manually or

electronically operated. Electronic operation may be in response to a polling operation carried out between all the communication devices in the group.

Preferably the signals received at the master are adjusted gainwise according to the strength of the weakest signal. This allows all of the signals to be heard, and in order to ensure that the strongest signals are heard more clearly than the weaker signals, the sounds output by the communication devices are permitted to vary according to the strength of the respective received signal.

In further embodiments of the invention a tone is added to signals transmitted by the master and the tone is used by the slaves to correlate with the master. A second tone is added by each slave to a signal sent to the master, the signal sent to the master is echoed directly back to the slave and the time delay measured. From the measured time delay it is possible to time transmissions from the respective slave so as to compensate for the delay time. Thus the signals received at the master are all synchronized and the master is able to correlate with the slaves, despite the fact that there are many of them.

It should be realized by one who is skilled in the art, that the above described many-to-many intercom system, can be adapted for a one-to-many half-duplex communication usage, simply by adding a VOX (Voice Operated Control) or a PTT (Push To Talk) switch to each of the slave devices.

Brief Description Of The Drawings

For a better understanding of the invention and to show how the same may be carried into effect, reference will now be made, purely by way of example, to the accompanying drawings in which,

Figure 1 is a block diagram showing the general principle of the first embodiment of the present invention;

Figure 2 is a block diagram showing the general principle of the second embodiment of the present invention;

Figure 3A is a block diagram of the CDMA/TDD/DSB (CDMA=Code Division Multiple Access, TDD=Time Division Duplex, DSB=Double Side-Band) and TDMA/TDD/DSB (TDMA=Time Division Multiple Access) modes of the first embodiment of the present invention, and the CDMA/TDD/DSB mode of the second embodiment of the present invention;

Figure 3B is a block diagram of the FDMA/TDD/DSB (FDMA=Frequency Division Multiple Access) mode of the first embodiment of the present invention;

Figure 3C is a block diagram of the CDMA/FDD/DSB (FDD= Frequency Division Duplex) and the TDMA/FDD/DSB modes of the first embodiment of the present invention; and the CDMA/FDD/DSB mode of the second embodiment of the present invention;

Figure 3D is a block diagram of the FDMA/FDD/DSB mode of the first embodiment of the present invention;

Figure 4A is a block diagram of the CDMA/TDD/SSB (SSB=Single Side-Band) and TDMA/TDD/SSB modes of the first embodiment of the present invention, and the CDMA/TDD/SSB mode of the second embodiment of the present invention;

Figure 4B is a block diagram of the FDMA/TDD/SSB mode of the first embodiment of the present invention;

Figure 4C is a block diagram of the CDMA/FDD/SSB and the TDMA/FDD/SSB modes of the first embodiment of the present invention, and the CDMA/FDD/SSB mode of the second embodiment of the present invention;

Figure 4D is a block diagram of the FDMA/FDD/SSB mode of the first embodiment of the present invention;

Figure 5A is a block diagram of the Logic block 20 of the CDMA/TDD, TDMA/TDD, CDMA/FDD and TDMA/FDD modes of the first embodiment of the first and second aspects of the present invention;

Figure 5B is a block diagram of the Logic block 20, including a Correlator device 69, of the CDMA/TDD, and CDMA/FDD modes of the second embodiment of the first and second aspects of the present invention;

Figure 5C is a block diagram of the Logic block 20 of the FDMA/TDD mode of the first embodiment of the first and second aspects of the present invention;

Figure 5D is a block diagram of the Logic block 20 of the FDMA/FDD mode of the first embodiment of the first and second aspects of the present invention;

Figure 6A is a truth table illustrating the operation of the components of Figures 5A; 5B and 5C;

Figure 6B is a truth table illustrating the operation of the components of Figures 5D;

Figure 7 is a diagram showing how the devices of Figures 3A; 3B; 3C and 3D, and Figures 4A; 4B; 4C and 4D may interrelate;

Figure 8A is a timing diagram showing how the devices of Figure 7 may communicate for the CDMA/TDD and TDMA/TDD modes of the first embodiment of the first and second aspects of the present invention, and for the CDMA/TDD mode of the second embodiment of the first and second aspects of the present invention;

Figure 8B is a timing diagram showing how the devices of Figure 7 may communicate for the FDMA/TDD mode of the first embodiment of the first and second aspects of the present invention;

Figure 8C is a timing diagram showing how the devices of Figure 7 may communicate for the CDMA/FDD and TDMA/FDD modes of the first embodiment of the first and second aspects of the present invention, and for the CDMA/FDD mode of the second embodiment of the first and second aspects of the present invention;

Figure 9A is a timing diagram illustrating the effects of distance on the devices of Figure 7 for the CDMA/TDD, TDMA/TDD, CDMA/FDD and TDMA/FDD modes of the first embodiment of the first and second aspects of the present invention;

Figure 9B is a timing diagram illustrating the effects of distance on the devices of Figure 7 for the CDMA/TDD and CDMA/FDD modes of the second embodiment of the first and second aspects of the present invention;

Figure 10 is a spectral analysis of the signal at the output of the master's correlator device 12, for the CDMA/TDD, TDMA/TDD, CDMA/FDD and TDMA/FDD modes of the first embodiment of the first and second aspects of the present invention;

Figure 11A is a spectral analysis of a transmission signal at the output of the devices of Figure 7, for the CDMA/TDD, TDMA/TDD, CDMA/FDD and TDMA/FDD modes of the first embodiment of the first and second aspects of the present invention, and for the CDMA/TDD and CDMA/FDD modes of the second embodiment of the first and second aspects of the present invention;

Figure 11B is a spectral analysis of a transmission signal at the output of the devices of Figure 7, for the FDMA/TDD mode of the first embodiment of the first and second aspects of the present invention;

Figure 12 is a spectral analysis of the result of interlacing 3 sets of signals of the type shown in Figure 7 for the CDMA/TDD and CDMA/FDD modes of the first embodiment of the first and second aspects of the present invention, and for the CDMA/TDD and CDMA/FDD modes of the second embodiment of the first and second aspects of the present invention;

Figure 13A is a detailed block diagram showing the relationship between all of the devices of Figure 7 for the CDMA/TDD, TDMA/TDD, CDMA/FDD and TDMA/FDD modes of the first embodiment of the first and second aspects of the present invention;

Figure 13B is a detailed block diagram showing the relationship between all of the devices of Figure 7 for the FDMA/TDD and FDMA/FDD modes of the first embodiment of the first and second aspects of the present invention, and for the CDMA/TDD and CDMA/FDD modes of the second-embodiment of the first and second aspects of the present invention;

Figure 14 is a diagram showing the effects of distance on the operation of devices according to the CDMA/TDD, TDMA/TDD, CDMA/FDD and TDMA/FDD modes of the first embodiment of the first and second aspects of the present invention.

Description Of The Preferred Embodiments

Figure 1 is a generalized diagram of an arrangement of a radio intercom system illustrating the principle of the first embodiment of the present invention. A series of slave transmit-receive units 202a-202n are located at different distances from a master transmit-receive unit 204, to which they are all synchronized. That is to say transmissions from the master to each slave are synchronized at the slave. The master unit 204 comprises a receiver 206, a pseudo random generator 208 a phase shifter 210 and a LO 218. The output of the pseudo random generator 208 and the LO 218 are fed to the phase shifter 210, and the outputs of the phase shifter 210 and of the receiver 206 are supplied to a demodulator, which is shown here as a product detector 212. The input to the receiver is the sum 216 of the signals from the different slaves, and these are not mutually synchronized because the slaves are at random distances from the master 204. The phase shifter 210 converts the constant phase of the signal of the LO 218 to a signal with pseudo random phase which is supplied to the demodulator 212. The angle of phase shift is constantly changed by the pseudo random generator 208 and in effect, all of the slaves are demodulated and can be heard at output 214. How this is done is explained in more detail below.

Figure 2 shows the second embodiment of the present invention. Again a series of slave units 202a..202n are located at random distances from a master unit 204, and between them they form a radio intercom system. The slave units 202a..202n are synchronized to the master but the master is again unable to synchronize with all of the slaves 202a..202n as they are located at different distances from the master 204. Thus each slave generates a synchronization tone at a certain frequency assigned to that slave. The tone is transmitted to the master and echoed back to the slave. From the echo the slave is able to measure the time delay of the double journey from the slave to the master and back to the slave again. The slave is then able to set a time-lead in transmission to the master in accordance with that measured delay. If the above procedure is repeated by all of the slaves then all of the signals received by the master will arrive in phase and the master will be able to synchronize to a multitude of slaves at different distances therefrom. If the slave keeps resetting

its transmission time-lead throughout the duration of the communication then the slave can move with respect to the master during the call and remain synchronized.

In both of the above embodiments of the present invention there is thus provided a means of having multiple radio communication units in contact with a single master communication unit, all using the same communication channels and wherein the master does not vary in size or complexity from the slave units. Thus each unit may comprise all of the electronics to be both a master and a slave, and may be set to be one or the other by a simple switching operation. Thus the failure of the master does not lead to failure of the system, as any other unit can be selected as master in its place. The selection of the master may be manual, or in a preferred embodiment may be carried out automatically as part of the communication protocol between the units. As there is no need for the master to transmit on different channels to each of the slaves it can be as mobile as any of the slave units.

The two embodiments of the invention will now be explained in greater detail with respect to Figures 3 to 14.

Figure 3A is a block diagram of the CDMA/TDD/DSB mode and the TDMA/TDD/DSB mode of the first embodiment of a device 2 according to the first aspect of the invention. Figure 3A is also a block diagram of the CDMA/TDD/DSB mode of the second embodiment of a device 2 according to the first aspect of the invention. The device 2 is a mobile station of an intercom system, and may serve both as a master station and as a slave station. The master station replaces the base station in the first of the conventional intercom systems mentioned above, that is to say it serves as the station with which each other device communicates. It receives all of the transmissions from each slave station. It combines all of the transmissions and it transmits the combined transmissions, as a single signal, to every slave station. Thus each slave station transmits exclusively to the master and receives exclusively from the master. The intercom system therefore comprises a star connected system.

It is noted that in all modes of operation of the present invention, a "combination" operation, as opposed to a "mixing" operation, is performed at the master station. A combining operation means that the signals are combined linearly producing one set of sideband frequencies. A mixing

operation is a non-linear operation resulting in a series of harmonic and intermodulation frequencies.

In Figure 3A, an embodiment using double sideband modulation is disclosed and includes a single antenna 5. Connected to the antenna 5 is a RF BPF (RF=Radio Frequency, BPF=Band Pass Filter) 6, followed by a SPDT T/R (SPDT=Single Pole Double Throw, T/R=Transmit/Receive) toggle 8. The T/R toggle 8 is shown in the receive position, logical "0" state. The received signal is passed from the T/R toggle 8 to a LNA (Low Noise amplifier) 10 and thence to a device 12. In the CDMA/TDD, TDMA/TDD, CDMA/FDD and TDMA/FDD modes, the device 12 functions as a Correlator/Despreader unit. The device 12 receives the output of the PRNG-Rx (PRNG=Pseudo Random Number Generator, Rx=Receive) line 18, which is part of Logic block 20, so as to be able to despread the received signal. Logic block 20 generates the control signals, such as timing signals and other reference signals, needed by the rest of the device.

In the FDMA/TDD modes the device 12 functions as an analog multiplier 13, Figures 3B and 4B.

As shown in figures 5A - 5D, an oscillator 22 functions in a loop, with pre-scaler input line 31; pre-scaler divider 62; divider 64, PRNG block 60, PRNG-Rx line 18; correlator 12 and a control toggle 30, to provide a sliding correlator. At the same time, oscillator 22 functions as a VCO (Voltage Controlled Oscillator) in a PLL (Phase Locked Loop), with pre-scaler input line 31; pre-scaler divider 62; PD (Phase Detector) 63, loop filter 65, PLL enable/disable toggle 67 and a Vc (Control Voltage) line 33. During synchronization, LD (Lock Detect) line 75 is low hence toggle 67 is closed and the PLL is locked to the reference 58 through reference line 59. In order that the correlation process between the slave and the master will have a short and predetermined time, it is necessary to shift the slave oscillator frequency relative to the master oscillator frequency. By doing so, the slave pseudo-random-sequence slides faster relative to the master sequence. This is achieved by toggle 25 which is governed by the M/S line 51 and causes a frequency shift of the slave reference oscillator 58 relative to the master reference. Once the slave receiver is synchronized to the signal, LD line 75 goes high, toggle 67 opens hence the slave PLL is open and the oscillator 22 becomes a SO (Synchronous Oscillator) which is synchronized to the incoming signal through toggle 30 and provides a LO signal to the modulator 46 and demodulator 28. The synchronizing loop is only required for the receive phase, hence a control toggle 30. Instead of a SO 22, a Costas

loop can be used, or any methods as are well known to those skilled in the art. US Patent 4,355,404 discloses a carrier recovery method using a SO, that allows drifting of the carrier to be tracked.

In this embodiment there is provided a phase shifter 32, which is only used in the master mode, hence it is connected to a M/S (Master/Slave) toggle 14. The reason that the master requires a phase shifter 32 is that the signals received from the slave stations are not synchronized. The entire system uses only a single frequency and since all of the slave stations are at different distances from the master, indeed since all of the stations are mobile, it is not possible for the master to be synchronized simultaneously to more than one of the slaves. It is possible for each of the slaves to synchronize to the master but not vice versa. Hence phase shifting is needed to receive an intelligible signal, as follows:-

Double sideband modulation signal gives:

$$S_{DSB}(t)=\cos(\omega_m t).\cos(\omega_c t)$$
$$=1/2\cos(\omega_c+\omega_m)t+1/2\cos(\omega_c-\omega_m)t$$

where ω_m is the message frequency, ω_c is the carrier frequency, and t is time.

In the case of synchronous double sideband demodulation,

$$S_{Sync}(t)=\cos(\omega_c t).[\cos(\omega_c t).\cos(\omega_m t)]$$
$$=[\cos(\omega_c t).\cos(\omega_c t)].\cos(\omega_m t)$$
$$=[1/2\cos(\omega_c-\omega_c)t+1/2\cos(\omega_c+\omega_c)t].\cos(\omega_m t)$$
$$=[1/2\cos 0+1/2\cos(2\omega_c t)].\cos\omega_m t$$
$$=1/2\cos(\omega_m t)\quad\text{(after low pass filtering).}$$

In the case of non- synchronous double sideband demodulation an additional term φ finds its way into the equation, as follows

$$S_{Non\text{-}sync}(t)=\cos(\omega_c t+\varphi).[\cos(\omega_c t).\cos(\omega_m t)]$$
$$=1/2\cos\varphi\cos(\omega_m t)\quad\text{(after low pass filtering).}$$

In order to render this signal intelligible it is necessary simply to use phase shifter 32 as one of the inputs to the demodulator 28. The phase shifter 32 pseudo randomly shifts the phase of the LO 22 signal to cancel out the effect of the "cos φ" term in the equation. It does this by producing an output which averages the "cos φ" term to $2/\pi$, that is to say the expression effectively becomes $S_{Non\text{-}sync}(t)=1/\pi\cos(\omega_m t)$. Thus a fully understandable signal is derived even though the receiver of the master remains unsynchcronized.

As the master is the only device that receives unsynchcronized signals it is only when the device is operating in master mode that the phase shifter 32 is needed. The output of the demodulator 28 is passed to the input of the Audio BPF 34. The Audio BPF 34 is comprised of two sections. One section is a band-limited 300 Hz to 3 KHz which is used to pass the audible spectrum of the received signal when the device 2 operates either as a master or as a slave. From this section the signal is passed to a volume controlled audio amplifier 36 and thence to a speaker 38. The second section of the Audio BPF 34 is a band-limited 50 Hz to 250 Hz which is used to pass the sub-audible spectrum of the received signal when the device 2 operates as a slave. From this section the signal is passed to logic block 20 through T_{in} line 35, and is used for synchronization and correlation purposes. How this is done is explained in more detail below.

The transmission part of the circuit is a conventional DSB transmitter circuit which preferably uses spread spectrum. An advantage of using spread spectrum is that interference and multi-path effects, which are generally frequency-dependent, are greatly reduced.

As shown in Figure 3A, a microphone 40 passes a speech signal to an Audio BPF 42, through the microphone AGC (Automatic Gain Control) 43. The Audio BPF 42 passes the signal, via an Audio amplifier 44, to a DSB/AM modulator 46 which modulates the speech signal on the carrier signal from the oscillator 22. The modulated signal is then passed to a Tx analog multiplier 48 which spreads the signal spectrum on the basis of the output PNRG-Tx (Tx=Transmit) line 17 of the pseudo-random generator 60, which is part of the logic block 20. The spread spectrum signal is then passed through two further RF amplifiers 50 and 52 and then transmitted via T/R toggle 8, RF BPF 6 and antenna 5.

Preferably the device 2 uses a single supply of DC (Direct Current) input 55, usually in the form of a battery. Unfortunately not all the blocks inside the device 2 can operate directly from this power source. Therefore a DC/DC converter and level-shifter 54 is provided in order to shift the level of the T/R line 77 to the level shifted T/R line 56, and also to provide the various DC voltages needed. In the embodiment shown, line 55 is 3V and line 56 is 5V, however as the state of the art progresses, these values may change.

It should be noted that BPF 34 and 42, Microphone AGC 43, amplifiers 36 and 44 and parts of the demodulator 28 modulator 46 and logic block 20 can be implemented either by HW (Hardware) or by a SW (Software) controlled DSP (Digital Signal Processor) as known in the art.

Figure 3B is a block diagram of the FDMA/TDD/DSB mode of the first embodiment of the first aspect of the invention. In the mode of Figure 3B, the PRNG-Tx line 17 and the PRNG-Rx line 18 of Figure 3A are replaced by DAC (Digital to Analog Converter) lines 19. One line is connected to an analog multiplier device 13 in the Rx path, and the second line is connected to an analog multiplier device 48 in the Tx path. A PRNG-Rx line 18 from Logic block 20 is connected to SPST (Single Pole Single Throw) toggle 14. This mode also includes two antennas 4.

Figure 3C is a block diagram of the CDMA/FDD/DSB mode and the TDMA/FDD/DSB mode of the first embodiment of the first aspect of the invention. Figure 3C is also a block diagram of the CDMA/FDD/DSB mode of the second embodiment of the first aspect of the invention. In the block diagram of Figure 3C, the T/R toggle 8, of Figure 3A, is replaced by a duplexer 9. Additionally the single frequency single port SO 22 is replaced by a dual-frequency dual-port SO block 23.

Figure 3D is a block diagram of the FDMA/FDD/DSB mode of the first embodiment of the first aspect of the invention. In the block diagram of Figure 3D, the T/R toggle 8, of Figure 3A, is replaced by a duplexer 9. Furthermore, the Rx Correlator 12 and Tx analog multiplier 48 are omitted and the PRNG-Rx line 18 is connected directly to toggle 14. As in the block diagram of Figure 3C, the single frequency single port SO 22 is replaced by a dual-frequency dual-port SO block 23.

Figure 4A is a block diagram of the CDMA/TDD/SSB mode and the TDMA/TDD/SSB mode of the first embodiment of a device according to the second aspect of the invention. Figure 4A is also a block diagram of the CDMA/TDD/SSB mode of the second embodiment of a device according to the second aspect of the invention. Figure 4A is a single sideband version of the device of Figure 3A. Parts that are the same as those shown in Figure 3A are given identical reference numerals. The most notable change between Figure 4A and Figure 3A is the SSB modulator 47 which is completely different from the DSB/AM modulator 46, whereas the demodulator 28 is the same for DSB AM and SSB. Using single sideband modulation the correlated signal that emerges from the Audio LPF (Low Pass Filter) is:

$$S_{Sync}(t)=\cos(\omega_m t).$$

If the signal is unsynchronized then the emerging signal is

$$S_{Non\text{-}sync}(t)=\cos(\omega_m t + \varphi),$$

that is to say it has an extra term "$+ \varphi$" which represents a random phase error due to the signal being unsynchronized. The human ear is insensitive to phase. Consequently the extra term "$+ \varphi$" makes no discernible difference and the signal can be understood without any further processing being necessary.

It will be realized that a number of features of conventional transmitter/ receivers are missing from the two block diagrams of Figures 3A - 3D and Figure 4A - 4D. One of them is a feedback link from the mouthpiece to the earpiece. The reason there is no such link is to ensure that the only way in which the user is able to hear himself is through the network. Hence the user has some way of being sure that his signal has been received. Such a feedback link only exists in the master device since the master user cannot hear himself through the network, hence M/S toggle 39 is provided.

Another feature that is missing is receiver AGC. Receiver AGC is used in radio and telephony to compensate for strong or weak signals and ensure that all are heard at the same volume. The absence of AGC ensures that the volume of the sound heard by the listener is linked to the received signal strength. Thus the user has a cue to tell him when he is getting too far away from the master and thus increases the volume of his voice. If the user is getting too close to the master he may decrease his volume. Thus everyone within the operating range of the system, as explained hereinbelow with respect to Figure 14, is heard with the same audio volume level. Additionally, the rest of the group can obtain an impression as to whether the user is nearby or far away, thus making the system more intuitive. Instead of AGC the signal received at the master is preferably adjusted to make the weakest signal audible, by a manual volume control 41.

Figure 4B is a block diagram of the FDMA/TDD/SSB mode of the first embodiment of the second aspect of the invention. In the mode of Figure 4B, the PRNG-Tx line 17 and the PRNG-Rx line 18 of Figure 4A are replaced by DAC lines 19. This mode also includes two antennas 4.

Figure 4C is a block diagram of the CDMA/FDD/SSB mode and the TDMA/FDD/SSB mode of the first embodiment of the second aspect of the invention. Figure 4C is also a block diagram of the CDMA/FDD/SSB mode of the second embodiment of the second aspect of the invention. In the

block diagram of Figure 4C, the T/R toggle 8 is replaced by a duplexer 9. Additionally the single-frequency single port SO 22 is replaced by a dual-frequency dual-port SO block 23.

Figure 4D is a block diagram of the FDMA/FDD/SSB mode the first embodiment of the second aspect of the invention. In the block diagram of Figure 4D, the T/R toggle 8 is replaced by a duplexer 9. Furthermore, the Rx Correlator 12 and Tx analog multiplier 48 are omitted. As in the block diagram of Figure 4C, the single-frequency single port SO 22 is replaced by a dual-frequency dual-port SO block 23.

Figure 5A shows in greater detail the Logic block 20 of the Figures 3A and 3C, and Figures 4A and 4C of the first embodiment of the first and second aspects of the invention. The purpose of the Logic block 20 is to provide the control signals needed for the operation of the units in Figures 3A and 3C, and Figures 4A and 4C. Reference numeral 60 denotes a pseudo-random generator at whose output 18 is provided the pseudo-random signal PRNG-Rx that is used for de-spreading the spectrum and for operating the phase shifter 32. Reference numeral 60 also provides output 17 which is the pseudo-random signal PRNG-Tx that is used for spreading the spectrum.

The items denoted by reference numerals 62, 64, 66 are dividers. The output of the second divider is used in this embodiment to clock the pseudo random generator 60. The output of PL-Encoder (PL=Private Line) 68 is a tone of predetermined frequency which is passed, via Audio LPF 70, and tone output enable/disable toggle 72, to comprise the signal T_{out} 37 that is added to the output signal. In this embodiment it is only passed on when the device 2 is operating as the master. The tone is at a frequency which is preferably inaudible to the human ear and the purpose of the tone is to provide a cue for the slaves that they are synchronized to the master. This is done in the slave by routing the T_{in} line 35 through the tone input enable/disable toggle 71 to the PL-decoder-LD block 74. In the embodiment shown in Figure 5A the tone is band limited at 250 Hz. Hence T_{out} line 37 is added to the voice signal downstream of Audio BPF 42 which is set to 300Hz - 3KHz, the generally recognized frequency dynamic range of the human voice.

The input M/S is the Master/Slave toggle 51, which may be set by a user, manually or automatically.

The input code selector 45 is used to select the different codes for the various groups in the CDMA modes. The input CDMA frequency interlace selector 53 is used to shift the master frequency reference 58, hence shifting its carrier frequency as will be described with reference to Figure 12 below.

In the various modes of the embodiments in Figures 3A -3D and Figures 4A - 4D, the dual-modulus pre-scaler divider 62 can be replaced by a frequency synthesizer. This is useful for example in some of the embodiments described below, in which provision is made for allowing several groups to operate together.

The frequency selector 49 may select either a divide by 128 or divide by 129 function in a dual-modulus pre-scaler divider 62. The output frequency is given as:

$F_o = 128 * F_{REF}$; or

$F_o = 129 * F_{REF}$,

where F_{REF} is the frequency output of a frequency reference oscillator 58. In order to expand this generic frequency synthesizer to a full frequency synthesizer, the dual-modulus pre-scaler divider 62 may be replaced by a divider scheme which may be controlled by a micro-processor, as is known in the art.

A M/S selector 51 is pulled-down to ground through a resistor 27. It is customary to tie logic lines either to ground (pull-down, ground = “0”) or pull-up to a supply voltage (pull-up, supply = “1”). The M/S selector remains in its current state until the state is changed by the M/S selector line 51.

The XOR (XOR= Exclusive Or) gates 76, 78 and the T/R enable/disable toggle 73 are used to operate the T/R line 77 according to the truth table of figure 6A. The NOR (NOR=Not Or) gate 79 is used to operate the control (Cont.) line 29 according to the truth table of figure 6A.

Figure 5B is a block diagram of the Logic block 20 of the Figures 3A and 3C, and Figures 4A and 4C of the second embodiment of the first and second aspects of the invention, in which a Correlator block 69 is added to the PL-decoder-LD block 74 and the tone output enable disable switch 72 is omitted, because in the second embodiment both master and slaves are transmitting sub-audible PL tones. The master transmits a PL tone for the slaves to synchronize, and each slave

transmits a different PL tone which is echoed by the master and used by the slave to time its transmission in order for the master to synchronize to all the slaves as explained hereinafter. The sequence of events is as follows: When a particular slave is synchronized to the master, T_{in} signal 35 is being detected by the PL-LD block 74 and the LD line 75 is pulled to high ("1") state. This in turn closes clock-shift enable/disable toggle 57. T_{out} PL signal 37, which is a different PL tone than the master originated PL tone, and is also a different PL tone for each slave, is transmitted by that particular slave, being echoed by the master as T_{in} signal 35, enters the correlator block 69, and causes the PRNG block 60 to time-shift the PRNG-Tx sequence on the PRNG-Tx line 17 relative to the PRNG-Rx sequence on the PRNG-Rx line 18. The time shift is such that at maximum correlation of the T_{in} signal 35, the PRNG-Tx sequence leads in time the PRNG-Rx sequence in an amount which is exactly the propagation time from that particular slave to the master and back, hence the master is fully synchronized to that particular slave. Since this is done by each slave, the master is synchronized to all of the slaves despite the fact that the slaves are at different distances from the master.

It should be noted that in the second embodiment, the T_{in} line 35 passes two different PL tones. One is originated by the master and used by the PL-LD block 74, and the second is originated by the slave , echoed by the master and used by correlator block 69.

Figure 5C shows in greater detail the Logic block 20 of the Figures 3B and 4B. The purpose of the Logic block 20 is to provide the control signals needed for the operation of the units in Figures 3B and 4B. In the device of Figure 5C, a Correlator block 69 is added to the PL-decoder-LD block 74. Additionally the PRNG-Tx line 17 is changed to a DAC line 19 by adding a DAC block 61.

The PL tone is used in the embodiment shown. An advantage of using the tone for correlation purposes is that the information signal can then be shaped, by the DAC block 61, using a Gaussian curve. An information signal shaped in this way produces a sharper spectrum and thus is a more efficient user of bandwidth. In the FDMA/TDD embodiments, the same PL tone originated by the master is routed on the T_{in} line 35 through toggle 71 to both the PL-LD block 74 and the correlator block 69. In these embodiments the synchronization process comprised of two steps. First step, the carrier of the slave is synchronized to the carrier of the master, using the sliding correlator mechanism as described with reference to Figure 3A above, which is recognized by the PL-LD

block 74 which pulls the LD line 75 high, hence closing toggle 57. Second step, the correlator block 69 correlates to the PL tone causing the PRNG block 60 to time-shift the TDD shaped pulses on the DAC line 19, in such a manner as to synchronize the TDD timing of the slave to the TDD timing of the master.

Figure 5D shows in greater detail the Logic block 20 of the Figures 3D and 4D. The purpose of the Logic block 20 is to provide the control signals needed for the operation of the units in Figures 3D and 4D. In Figure 5D the PRNG-Tx line 17 is omitted, and the PRNG-Rx line 18 is only used in the first aspect of the invention as an input to the phase shifter block 32.

Figure 6A is a truth table, illustrating the operation of the Logic block 20 illustrated in Figures 5A, 5B and 5C. Figure 6A shows the dependence of output signals Cont. line 29, Vc line 33 and T/R line 77 on the M/S line 51 and the LD status line 75.

Row-1 in Figure 6A represents a stable state of a master and row-3 represents a stable state of a slave when in synchronization with the master. Row-2 represents a metastable state of the slave before going into a synchronized state (row-3).

Figure 6B is a truth table, illustrating the operation of the elements of Figure 5D for the FDMA/FDD/DSB and FDMA/FDD/SSB modes.

Returning to the embodiment using a tone sent by the master for correlation purposes, it is also possible to provide the slave devices with timing means. It will be recalled that the signal sent by the slave is received back in the return signal. The slave can include its own tone in the signal sent to the master and "measure" the delay. The slave can use this information to time the sending of its own signal to the master so as to compensate for the delay due to the distance. If all of the slaves follow this procedure then all of the signals received at the master will be in synchronism and the master will be able to synchronize to all the received signals.

The advantage of the present method, over the above described previous methods, is that as all of the units are fully synchronized a lower power signal can be used, or alternatively a signal of similar power will have a greater range. Nevertheless such a system is more sensitive to interference, particularly multiple path interference, which can make correlation to the PL tones difficult. Thus in a preferred embodiment the communication devices are able to switch from

synchronizing in the above-described manner to transmitting unsynchronized signals as described with reference to Figures 3A - 3D and 4A - 4D above.

In use several of the devices 2 form a group. One of the devices is configured as the master and the rest are the slaves, as shown in Figure 7. TDD is used to divide the time between transmission by the master and transmission by the slaves, as shown in Figures 8A and 8B for the various modes of the embodiments. Each slave transmits exclusively to the master, and it preferably transmits only the signal from its microphone. The master receives transmissions from any slave that happens to transmit in any particular slave transmit period, and combines them together to transmit to all the slaves during the master transmit period, together with any voice signal added at the master device itself. As discussed above, the slave devices all synchronize to the master during the master transmit period. However the master cannot attempt to synchronize to the slaves during the slave transmit period because the slaves are all at different distances from the master, and thus their signals are offset slightly from one another.

Figure 8A shows the timing diagram for the communication between the master and slave devices of Figure 7, for the CDMA/TDD modes. Figure 8A shows that the duplex period T_D is equal to the pseudo-random-sequence period T_P. In the CDMA/TDD and the TDMA/TDD modes, T_D is chosen to satisfy the Nyquist frequency in order that the audio signals can be reconstructed using simple filter. The audio frequency is limited by the Tx audio filter 42 to 3 KHz, hence the Nyquist frequency is 6 KHz and the T_D is chosen to be 7 KHz in the above embodiments, and is governed by the T/R line 77. The reconstruction of the audio signals are done by the Rx audio filter 34. The synchronization between T_D and T_P is done by identifying a known portion of the pseudo-random-sequence. In the embodiments shown, this part is the 'all 1' portion, which is identified by the PRNG block 60 and used to reset the counter/divider 66.

It should be noted that while a slave is acquired carrier and pseudo-random-sequence synchronization, that slave is in the Rx only state, or metastable state as depicted in figure 6A. When that slave is carrier and pseudo-random-sequence synchronized, its T/R line 77 toggles between Tx and Rx states in an manner opposite to that of the master, as depicted in figure 6A, hence the TDD is synchronized.

Figure 8A shows a 50% duty cycle on the T_D period which is the situation in the CDMA/TDD modes, i.e. Tx and Rx periods are 50% each of the T_D time. In the TDMA/TDD modes the duty cycle is 50% divided by the number of slaves in the group. For example, if there are 5 slaves in the group, the duty cycle is 10% which means that each slave transmits 10% of the T_D time and then receives 10% of the T_D time. The synchronization of T_D and T_P are done in the same way as described above for the CDMA/TDD modes.

The duplex period is preferably equivalent to the code period for the system being used, for example a 128 chip code may be used. The slave transmit period may then be 64 chips and the master transmit period 64 chips. If the code period were 127 chips then one of the two transmit periods would have to be one chip longer than the other. The code is chosen for minimum cross-correlation over each transmit period, as opposed to the entire code period. This is for the benefit of the first slave which joins the group after initialization. This slave is required to synchronize whilst only being able to see half the code.

Figure 8B is a timing diagram showing how the devices of Figure 7 may communicate for the FDMA/TDD modes. The carrier synchronization is achieved using the mechanism as described in US Patent 4,355,404 above, which is recognized by the PL-LD block 74. The TDD synchronization is then achieved using the correlator block 69 and the PL tone transmitted by the master as described with reference to Figure 5C above.

Figure 8C is a timing diagram showing how the devices of Figure 7 may communicate for the CDMA/FDD and the TDMA/FDD modes. Carrier and code synchronization is achieved using the sliding correlator mechanism as described with reference to Figure 3A above. TDMA synchronization is achieved using the 'all 1' detection as described with reference to Figure 8A above.

The above-described embodiments thus provide a fully duplex intercom system that is able to accommodate a group of users of reasonable size using just a single duplex channel. In addition the master device is fully portable and differs from the slave device only by means of a few circuit elements. It is thus possible to provide devices that have all the circuit elements needed for both master and slave mode and to configure the device accordingly. Thus the group is not dependent on the serviceability of a single master device. In the event of failure of the master it is necessary

simply to select another master. The selection of the master can be carried out manually by the users. In a preferred embodiment the appointment of a master is carried out automatically by electronic polling of the members of the group upon initialization, and if necessary later on as part of a re-initialization procedure following loss of signal from the original master.

Figure 9A is a snapshot of the master and slave Tx and Rx pseudo-random-sequences at a certain distance between the master and the slave, and shows the propagation delay effect upon signals being transmitted from master to slave and back to master again. The slave Rx sequence lags the master Tx sequence by an amount of time equals to the propagation delay which is dependent on the master-slave distance. Since the slave is fully synchronized on the arriving signal, the slave Tx is synchronized to the slave Rx sequence. The master Rx sequence lags the master Tx sequence by twice the propagation delay, and the amount of correlation between the master Tx and Rx sequences is the amount of overlapping of the corresponding chips. Figure 9A shows the correlated and uncorrelated parts of chip #4 of the master Tx and Rx sequences, but the same applies to the other chips. Correlation distance can be defined now as: $L_{corr}=(C*T_C)/2=(C/C_R)/2$ where T_C is the chip time length, $C_R=1/T_C$ is the chip rate and C is the speed of light. Correlation distance is the distance at which the correlated part of the corresponding chips are zero and the uncorrelated part equal the timelength of one chip.

Figure 9B shows the same details for the second embodiment, in which PL tones are used to align the two signals as described with reference to Figure 5B above. Information gleaned from the tones is used to set a clock lead between the slave Tx and Rx sequences, which is twice the propagation delay between the master and slave, so that the master Rx sequence is fully correlated to the master Tx sequence.

It should be noted that in the second embodiment there is no correlation distance since the master is always fully synchronized which each one of the slaves.

Figure 10 is the Fourier transform of Figure. 9A and shows the relationship between correlated and uncorrelated parts at the output of the correlator 12 of the master side receiver. It shows a spectral analysis of the signal after it has been despread. The signal-power relationships in these two parts are as follows,

for the correlated part, $S_O = ((L_{corr} - L_{dist})/L_{corr}) * S_I$

and for the uncorrelated part, $S_O = (L_{dist}/L_{corr}) * (S_I/G_P)$

where L_{dist} is the actual distance between the master and the slave, L_{corr} is the correlation distance, as defined with reference to Figure 9A above, S_I is the power of the original-unspread signal in the nominal information BW (Band-Width), S_O is the power of the signal in the nominal information BW at the output of the master's correlator 12, and G_p is the DSSS (Direct Sequence Spread Spectrum) process gain of the system.

It can be seen that the correlated part of the signal is a single spectral line at F_0, while the uncorrelated part comprises multiple spectral lines at F_0 and around F_0, having a Sinc(x) envelope.

It should be noted that when $L_{dist}>L_{corr}$, then there is no correlated part in S_O and the output of the master's correlator 12 is completely uncorrelated. Therefore the spectrum is the Fourier transform of the pseudo-random-sequence which is comprised of multiple spectral lines at F_0 and around F_0, having a Sinc(x) envelope, and the signal power is given by: $S_O=S_I/G_P$.

Figure 11A is a close look at the spectrum of a transmission signal at the output of the devices of Figure 7, for the CDMA and TDMA modes. Figure 11A is also a close look at the spectrum of the uncorrelated part of the signal of Figure 10 at the output of the master correlator 12. From the inset of Figure 11A it can be understood that even in the case that $L_{dist} >L_{corr}$ and the output of the master correlator 12 is completely uncorrelated, the information around F_0 can be demodulated using the product-detector demodulator 28 and the filter 34, provided that ΔF is greater than the information BW. In Figure 11A ΔF is shown as 7KHz, whereas the information BW is 6KHz, and is defined by the Fourier transform as: $\Delta F=1/T_P$, where T_P is the pseudo-random-sequence period.

It should be noted that the slaves are fully synchronized and therefore fully correlated. This is also the case regarding the master in the second embodiment. In these cases the output of the correlator 12 is a single spectral carrier C_0 at F_0 with 2 sidebands in the first aspect and either LSB (Lower Side-Band) or USB (Upper Side-Band) in the second aspect of the invention.

Figure 11B is a spectral analysis of a transmission signal at the output of the devices of Figure 7, for the FDMA/TDD modes. Figure 11B is also a spectral analysis of a transmission signal at the output of the devices of Figure 7 for the TDD portion of the CDMA and TDMA modes. From Fourier transform we get $\Delta F_{Duplex}=1/T_D$, and since $T_D=T_P$ for the CDMA/TDD and TDMA/TDD

modes as shown in Figure 8A, and since $\Delta F=1/T_P$ as described with reference to Figure 11A above, it follows that $\Delta F_{Duplex}=\Delta F$. Hence in the CDMA/TDD and TDMA/TDD modes the spectral lines coincide and there are no intermodulation products, so that the information can be demodulated as described with reference to Figure 11A above.

Preferably, as described with reference to Figure 8A above, the code should be chosen to have an even number of chips e.g. 128 where 64 chips are used by the master and 64 by the slave. The spectrums of both master and slave are identical as shown in Figure 11B/second graph. If the code is chosen to have an odd number of chips, e.g. 127 where 64 chips are used by the master and 63 by the slave, then the spectrum of the master which is shown in Figure 11B/first graph is different from the spectrum of the slave which is shown in Figure 11B/third graph.

Figure 12 shows an interlaced spectrum situation applicable to the CDMA modes. When more than one group wants to operate in the same vicinity, and the process gain G_P of the system is not high enough to overcome the 'Near-Far' problem as known in the DSSS art, then ΔF should be chosen to be a multiple of the Nyquist frequency plus some guard band. In Figure 12, ΔF~27KHz. From this it follows that more than one group, in this case 3 groups, can operate in the same vicinity. This is done simply by shifting the carrier frequency F_0 of the second group up by 9KHz and shifting the carrier frequency F_0 of the third group up by 18KHz. This way several groups can operate without interfering with one another. This may be achieved by providing the master with a circuit for identifying other active groups in the vicinity. Upon initialization, the master in each group identifies the presence of other groups and adjusts its carrier frequency F_0 accordingly.

It can be shown that $\Delta F=C_R/C_P$ where C_P is the length of the pseudo-random-sequence code, which indicates that ΔF can be increased by either increasing C_R or decreasing C_P. In the first embodiment the correlation distance $L_{corr}=(C/C_R)/2$ as described with reference to Figure 9A above. Therefore there is a limit to the extent of which C_R can be increased without impairing severely the correlation distance and hence the operating range of the system. In this case it is preferred to use a shorter code C_P. In the second embodiment there is no correlation distance, hence one can use a relatively large code C_P and increase the chip rate C_R in order to get large ΔF, which in turn enables to operate in an interlaced-spectrum mode and avoid the 'Near-Far' problem.

Figure 13A shows a model of a group consisting of a master and 'N' slaves for the CDMA and TDMA modes of the first embodiment of the invention. This model is used to analyze and to optimize the performance of the system. The slave microphone 80 receives the voice and the noise inputs of its user, which are controlled in amplitude by the slave microphone AGC 82 in conventional fashion. It is then transmitted as a spread spectrum signal via the slave transmitter 84. The channel 85 acts as an adder and the signals of all of the slaves are simply added together by the adder 83, making it impossible for the master to correlate. The noise output of all the slaves are also added by the adder 81.

It should be noted that the channel 85 is not a regular channel. it operates differently on the signal and on the noise. In addition to the regular signal losses of FSPL (Free Space Path Loss) and fades, there is also a signal loss due to the fact that the master is not correlated with the slave and the amount of decorrelation is a function of the distance between the master and slave L_{dist}, as described with reference to Figure 9A above. On the other hand, since the noise is modeled as AWGN (Additive White Gaussian Noise), there are no correlation effects on the noise as shown in Figure 13A.

It should also be noted that the noise is given a special consideration in this model, since the model deals with group simultaneous communication and hence the minimum S/N (Signal to Noise) at which the system can operate depends on the distance as well as the number of slaves, whereas in a regular communication system, the S/N depends only on the distance. This is due to the fact that each slave adds noise to the system.

The master receiver 86 receives the signal plus noise and does not need to correlate, as described above. The noise from the slaves is added at the input of the master to the master input thermal noise by adder 89. The signal plus noise are passed to the earpiece or speaker 88 of the master. Any speech plus noise added by the user of the master is received by the master microphone 90 and added to the received signal plus noise from the master receiver 86 by the master audio combiner 92. For the purpose of master self-hearing, the speech plus noise coming from the master microphone 90 are sampled and fed back to the master speaker 88 through the audio splitter/combiner 91. The combined signal and combined noise are amplitude controlled by the master microphone AGC 94 and then are broadcast to each of the slaves via the master transmitter

96. Since the slaves correlate to the master, the channel from master to slave is a regular communication channel 87 as seen in Figure 13A.

The signal plus noise are received by the slave receiver 98. The noise from the master is added at the input of the slave to the slave input thermal noise by adder 89. The signal plus noise are then passed to the slave earpiece or speaker 100. It will be noted that there is no direct connection between the slave earpiece 100 and the slave microphone 80. The reason for this, as explained above, is to allow the user a means of judging how well he can be heard by the rest of the group, and to allow the users to know how near or far others are from the master, that is to say it is a means of giving the group an intuitive feel.

Figure 13B shows a model of a group consisting of a master and 'N' slaves for the FDMA modes of the first embodiment, and for the CDMA modes of the second embodiment of the invention. Figure 13B is a regular channel version of the model of Figure 13A. Parts that are the same as those shown in Figure 13A are given identical reference numerals. These modes are characterized by the fact that the master is correlated to the slaves, hence both channels i.e. slave to master channel and master to slave channel behave as a regular communication channel 87.

Figure 14 illustrates how an intuitive group of this nature may work. A master 110 defines the center of a group. A first slave 112 is situated at a first radius 'a' from the master 110. Within this radius 'a' the slave 112 is able to hear itself, although the volume recedes as the distance increases. The user is able to compensate by speaking louder, but nevertheless becomes aware that he is reaching the limits of the self hearing radius 'a'. The circle zone which is defined by radius 'a' is called the S-H-I (Slave Hear Itself) zone. A second slave 114 is located at a radius 'b' from the master. The slave 114 can be heard by the network and the slave 114 can hear the network, although the sound fades as the radius 'b' is approached. However at this radius the slave 114 is unable to hear itself. The ring zone which is defined between radius 'a' and radius 'b' is called the M-H-S (Master Hear Slave) zone.

A third slave 116 is located at a radius 'c' from the master 110. At this distance the slave 116 is able to hear the master 110 and synchronize thereto, however the slave 116 cannot be heard by the network. Effectively, at this sort of range the slave 116 operates in a passive mode, providing

only a monitoring function. The ring zone which is defined between radius 'b' and radius 'c' is called the S-H-M (Slave Hear Master) zone. Beyond radius 'c', no communication is possible.

It should be noted that Figure 14 differs from a regular communication system at which only radius 'c' exists. The relationships among the radii 'a', 'b' and 'c' are determined by calculations based on the models of Figures 13A and 13B above, and are dependent also on the number of slaves, as described with reference to Figure 13A above.

A radio intercom system of this type has many uses. Because no large unwieldy base station is involved the system is entirely portable. As only a single channel is used the system is a lot cheaper than conventional radio intercom. Thus applications for which radio intercom was previously felt to be prohibitively expensive may now be contemplated. It can for example be used to maintain contact between teams of sportsmen in the field, or between mobile office staff or teams of engineers. It is particularly useful for looking after children. The responsible adult can have his communication device permanently set as master and will be made aware when any of his charges are wandering too far away.

Certain applications warrant particular adaptations of the way the communication device itself is made. For example cyclists or skiers will want a hands-free version preferably built into their helmets. This would also be suitable for teams of engineers. For use with children a format resembling a necklace may be more appropriate, as here too a hands-free version is useful but of course no helmet is being worn.

It is appreciated that the various features of the invention which are, for clarity, described in the contexts of separate embodiments may also be provided in combination in a single embodiment. Conversely, various features of the invention which are, for brevity, described in the context of a single embodiment may also be provided separately or in any suitable subcombination.

Claims

1. A method of simultaneously recovering a plurality of double sideband radio signals that are not mutually synchronized, comprising the steps of

receiving said plurality of double sideband radio signals that are not mutually synchronized, and

demodulating said plurality of double sideband radio signals that are not mutually synchronized using the output of a phase shifting device.

2. A method according to claim 1 in which said output of said phase shifting device is predetermined to have an average level which cancels an effect of a cosine term in said demodulated signal to render said plurality of received signals audible.

3. A method according to claim 1 or claim 2 wherein said phase shifting device produces an output which is proportional to a received input and wherein said phase shifting device receives an input which has a constant average level.

4. A method according to any one of claims 1 to 3 wherein said signals are amplitude modulated signals.

5. A method of simultaneously recovering a plurality of single sideband radio signals that are not mutually synchronized, comprising the steps of

receiving said plurality of single sideband radio signals that are not mutually synchronized, and

demodulating said plurality of single sideband radio signals that are not mutually synchronized.

6. A method according to any one of claims 1 to 5 wherein said radio signals are code division multiple access radio signals.

7. A method according to any one of claims 1 to 6 wherein said radio signals are time division multiple access radio signals.

8. A method according to any one of claims 1 to 7 wherein said radio signals are frequency division multiple access radio signals.

9. A method of sending an electronic signal from a first point to a second point remote from said first point and providing an indication to said first point regarding receipt of said signal at said second point, comprising the steps of relaying said electronic signal back from said second point to said first point and monitoring the level of said relayed signal at said first point.

10. A method according to claim 9 wherein said first point has a sound input site and a sound output site, wherein sound is converted into said electronic signal after passing through said sound input site, wherein said relayed signal is monitored by converting said relayed signal into sound at said sound output site and listening to said sound output site.

11. A method according to claim 10 wherein said electronic signal is passed to said sound output site only after being relayed from said second point.

12. A method according to any one of claims 9 to 11 wherein the level of the sound output from said sound output site is allowed to vary with said level of said relayed signal so as to reflect said average level of said signal.

13. A method according to any one of claims 10 to 12 wherein said second point has a sound output site to which said electronic signal is directed to for conversion into sound and wherein said level of said sound at said sound output site of said second point, resulting from said conversion, is allowed to vary with said level of said electronic signal so as to reflect said average level of said signal.

14. A method according to any one of claims 10 to 12 wherein said electronic signal is passed from said second point to at least one other point, wherein said at least one other point has a sound output site, wherein sound is output at said sound output site in response to said electronic signal and wherein said level of said sound output at said sound output site is allowed to vary with said level of said electronic signal so as to reflect said average level of said signal.

15. A method according to any one of claims 9 to 14, wherein said sound output by said first point is permitted to vary in strength according to the strength of a respective received signal.

16. Apparatus for simultaneously recovering a plurality of double sideband radio signals that are not mutually synchronized, comprising a demodulation device, the respective inputs of which are connected to a phase shifter device and to said radio signals.

17. Apparatus according to claim 16 wherein said phase shifter has an input connection, is adapted to produce a phase shift in accordance with a level of a signal received at said input connection, and wherein said input connection is connected to a device adapted to produce a signal having a constant average level.

18. An apparatus according to claim 16 or claim 17 wherein said input of said phase shifter is connected to said output of a pseudo random generator.

19. A method of providing wireless communication between a plurality of communication devices forming a group, such that each device receives transmissions from each other device in said group, comprising the steps of

selecting one of said devices as a master,

transmitting from said master to each other device using a preselected channel,

synchronizing each other device to said master,

transmitting from each other device to said master using said preselected channel,

combining said signals from each other device at said master, and

transmitting said combined signals as a single signal from said master to said each other device to be received at each other device.

20. A method according to claim 19 wherein transmission between said master and said other devices is carried out using time division duplex.

21. A method according to claim 19 or claim 20 wherein transmission between said master and said other devices is carried out using frequency division duplex.

22. A method according to any one of claims 19 to 21 wherein transmission between said master and said other devices is carried out using code division multiple access.

23. A method according to any one of claims 19 to 22 wherein transmission between said master and said other devices is carried out using time division multiple access.

24. A method according to any one of claims 19 to 23 wherein transmission between said master and said other devices is carried out using frequency division multiple access.

25. A method according to any one of claims 19 to 24, wherein said signals from said other devices are not mutually synchronized and wherein, upon receipt at said master, they are demodulated using a pseudo-random shifted local oscillator.

26. A method according to any one of claims 19 to 25 wherein failure of said master is followed by selection of another one of said communication devices to serve as master.

27. A method according to any one of claims 19 to 26, wherein at least one of said other devices has a sound input site and a sound output site, wherein sound is converted into an electronic signal after passing through said sound input site, wherein said electronic signal is transmitted to said master, wherein said signal received from said master includes said electronic signal sent to said master, and wherein said received signal is converted into sound at said sound output site.

28. A method according to any one of claims 19 to 27, wherein a tone is added to signals transmitted by said master and wherein said tone is used by at least one of said other devices to correlate with said master.

29. A method according to claim 28 wherein said signals transmitted with said tone are shaped using a Gaussian curve.

30. A method according to any one of claims 19 to 29, comprising the step of shifting a base frequency used for transmission upon detection by said master of other groups operating in the vicinity.

31. A communication device for use in the method of any one of claims 19 to 30, comprising circuitry for operation as one of said other devices and comprising additional circuitry for operation as said master, and further comprising a switching circuit for switching between said circuitry.

32. A communication device according to claim 31 wherein said switching circuit is manually operable.

33. A communication device according to claim 31 or claim 32 wherein said switching circuit is electronically operable.

34. A communication device according to claim 33 wherein said switching circuit is operable in response to a polling operation carried out between all said communication devices in said group.

35. A method according to any one of claims 19 to 34 wherein said signals received at said master are of variable strength, and are adjusted gainwise in accordance with said strength of a weakest signal.

36. A method according to any one of claims 19 to 29 and 35, wherein said sounds output by said communication devices are permitted to vary in strength according to the strength of a respective received signal.

37. A method according to claim 19 wherein a tone is added to signals transmitted by said master and wherein said tone is used by at least one of said other devices to correlate with said master, wherein a second tone is added by said one of said other devices to a signal sent to said master, wherein said signal is echoed back by said master to said one of said other devices, wherein said signal sent to said master is used by said one of said other devices to measure a delay time involved in transmitting to said master by auto-correlating said second tone, and wherein said measured delay time is used to time transmissions from said one of said other devices so as to compensate for said delay time.

38. A communication device adapted for use in the method of any one of claims 19 to 37.

39. At least three transmitter-receiver radio devices, wherein one of said devices is designated as master, wherein each other of said devices is adapted to transmit to said master over one channel of a dual channel link and to receive from said master over a second channel of said dual channel link, and wherein all of said devices are adapted to use said dual channel link.

40. At least two transmitter-receiver radio devices, wherein one of said devices is designated as master, wherein each other of said devices is adapted to transmit to said master over one channel of a dual channel link and to receive from said master over a second channel of said dual channel link, wherein each other of said devices has a microphone for sound input and a sound output device, wherein no operable connection is provided between said microphone and said sound output device of each other of said devices, and wherein operable connection is provided between said microphone and said sound output device of said master.

41. At least two transmitter-receiver radio devices, wherein one of said devices is designated as master, wherein each other of said devices is adapted to transmit to said master over one channel of a dual channel link and to receive from said master over a second channel of said dual channel link, wherein each device has a sound output unit, and wherein the volume of a sound output by said sound output unit is related to the strength of a received signal.

42. At least three transmitter-receiver radio devices, wherein one of said devices is designated as master, wherein each other of said devices is adapted to transmit to said master over one channel of a dual channel link and to receive from said master over a second channel of said dual channel link, wherein said master comprises a tone production device, and an adder connected to said tone production device, adapted to add a tone from said tone production device to any transmitted signal, and wherein at least one other of said devices is adapted to use said tone to correlate to said master.

43. At least three transmitter-receiver radio devices, wherein one of said devices is designated as master, wherein each other of said devices is adapted to transmit to said master over one channel of a dual channel link and to receive from said master over a second channel of said dual channel link, wherein at least one other of said devices comprises a tone production device, and an adder connected to said tone production device, adapted to add a tone from said tone production device to any transmitted signal, wherein said other device comprises a measuring unit for measuring a distance related delay in said transmission from a received tone received in said second channel by calculating an autocorrelation coefficient between said tone as transmitted and said tone as received, and a calculating unit for calculating a transmission time offset correction from said calculated autocorrelation coefficient, and wherein said at least one other device has an offset unit for re-timing transmissions in accordance with said transmission time offset correction.

44. At least three transmitter-receiver radio devices, wherein one of said devices is designated as master, wherein each other of said devices is adapted to transmit to said master over one channel of a dual channel link and to receive from said master over a second channel of said dual channel link, and wherein said master comprises a demodulating unit and a phase shifting unit.

45 A system of transmitter-receiver radio devices according to any one of claims 39 to 44 adapted to operate using code division multiple access.

46 A system of transmitter-receiver radio devices according to any one of claims 39 to 45 adapted to operate using time division multiple access.

47 A system of transmitter-receiver radio devices according to any one of claims 39 to 46 adapted to operate using frequency division multiple access.

48 A system of transmitter-receiver radio devices according to any one of claims 39 to 47 wherein said dual channel link is time division duplex.

49 A system of transmitter-receiver radio devices according to any one of claims 39 to 48 wherein said dual channel link is frequency division duplex.

50. A radio communication device substantially as hereinbefore described with reference to the accompanying drawings.

51. A method of providing wireless communication between a plurality of communication devices forming a group, such that each device receives transmissions from each other device in the group, substantially as hereinbefore described with reference to the accompanying drawings.

For the Applicant,

Sanford T. Colb & Co.
C: 27662

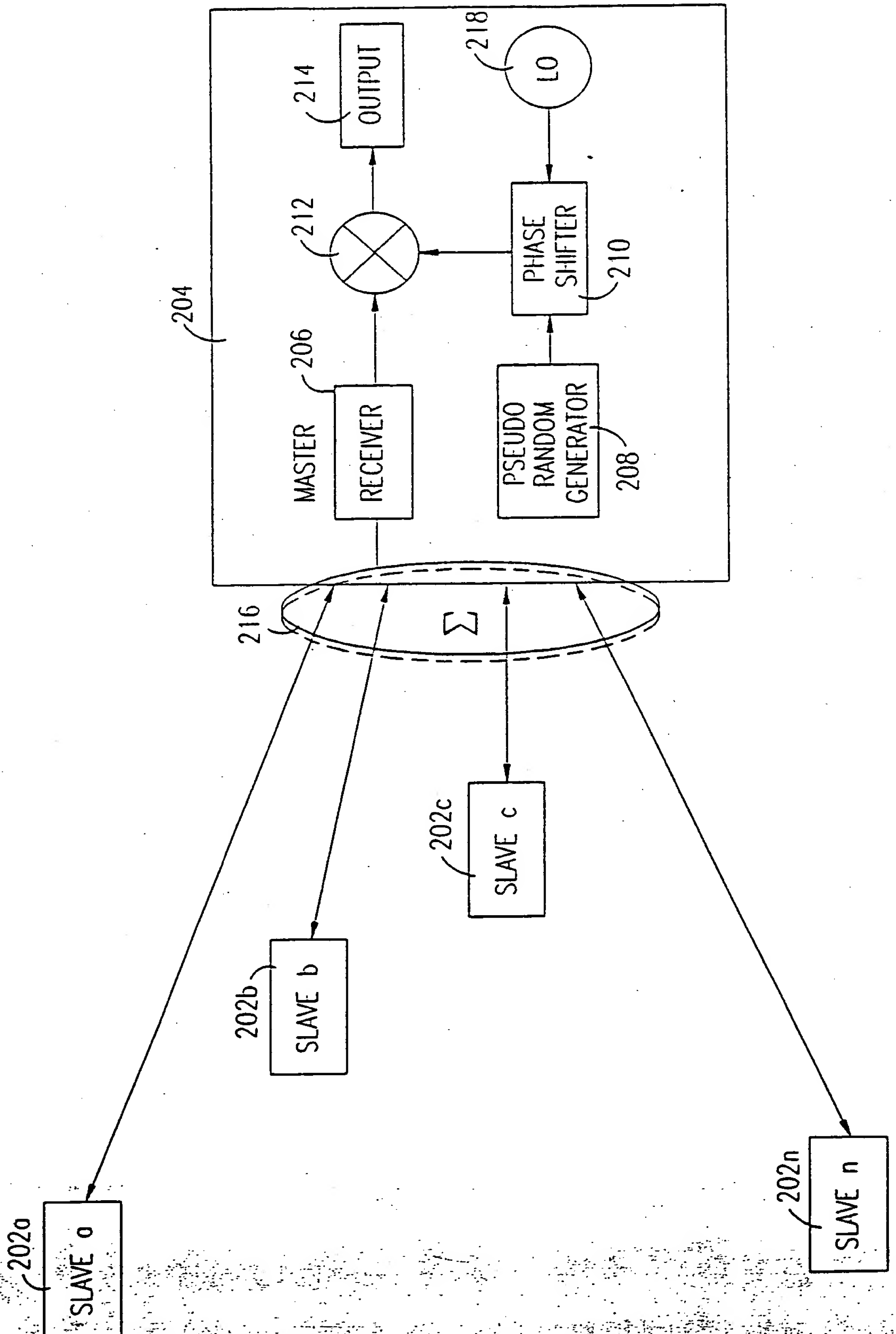

MASTER
RECEIVER
206
204
212
PHASE SHIFTER
210
OUTPUT
214
LO
218
PSEUDO RANDOM GENERATOR
208
216
Σ
SLAVE a
202a
SLAVE b
202b
SLAVE c
202c
SLAVE n
202n

FIG. 1

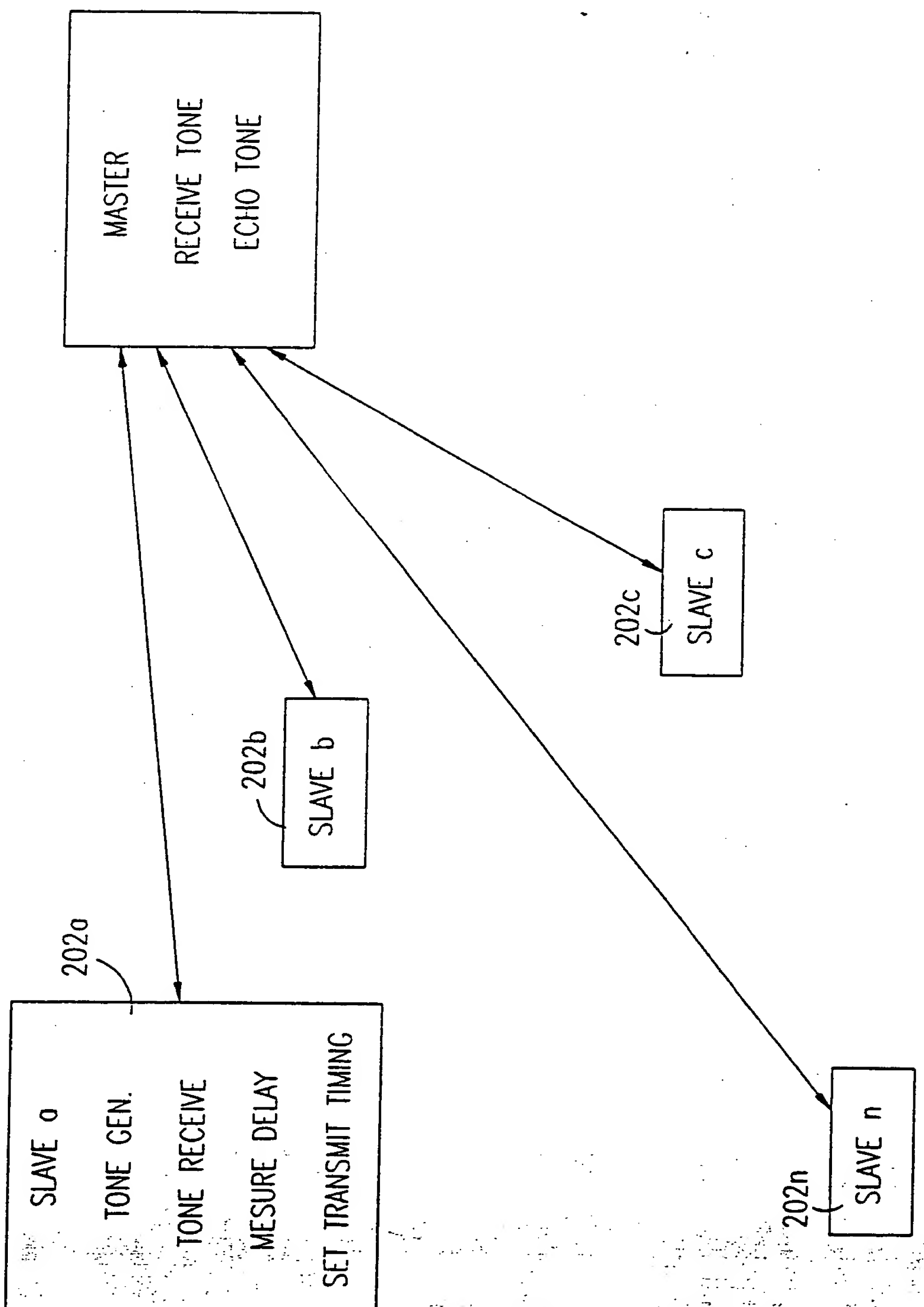

MASTER
RECEIVE TONE
ECHO TONE
SLAVE a
TONE GEN.
TONE RECEIVE
MESURE DELAY
SET TRANSMIT TIMING
202a
SLAVE b
202b
SLAVE c
202c
SLAVE n
202n

FIG. 2

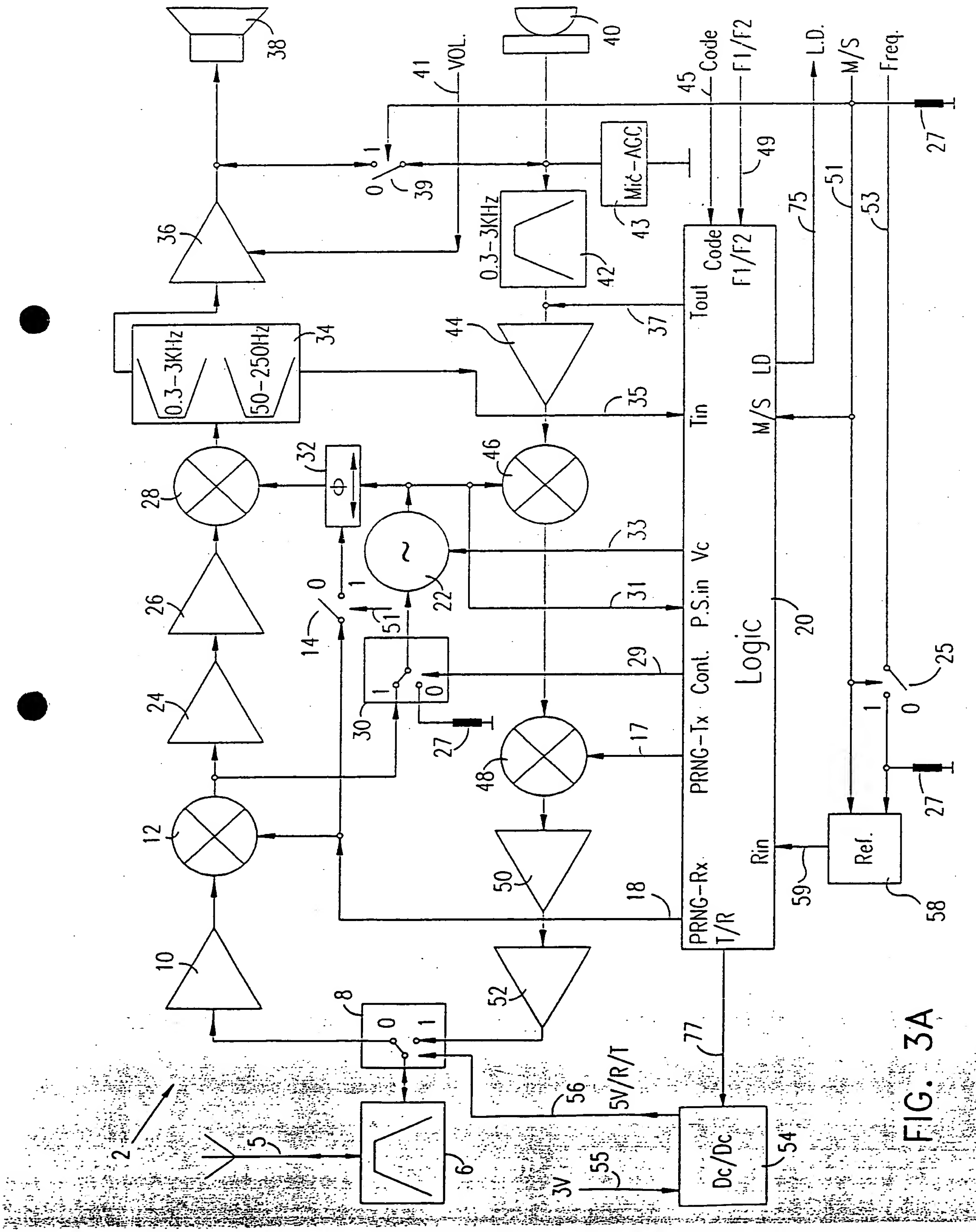

VOL.
Code
F1/F2
L.D.
M/S
Freq.
Mic-AGC
0.3-3KHz
50-250Hz
Tout
Tin
Vc
P.S.in
Cont.
PRNG-Tx
PRNG-Rx
T/R
Logic
LD
Rin
Ref.
Dc/Dc
5V/R/T
3V

FIG. 3A

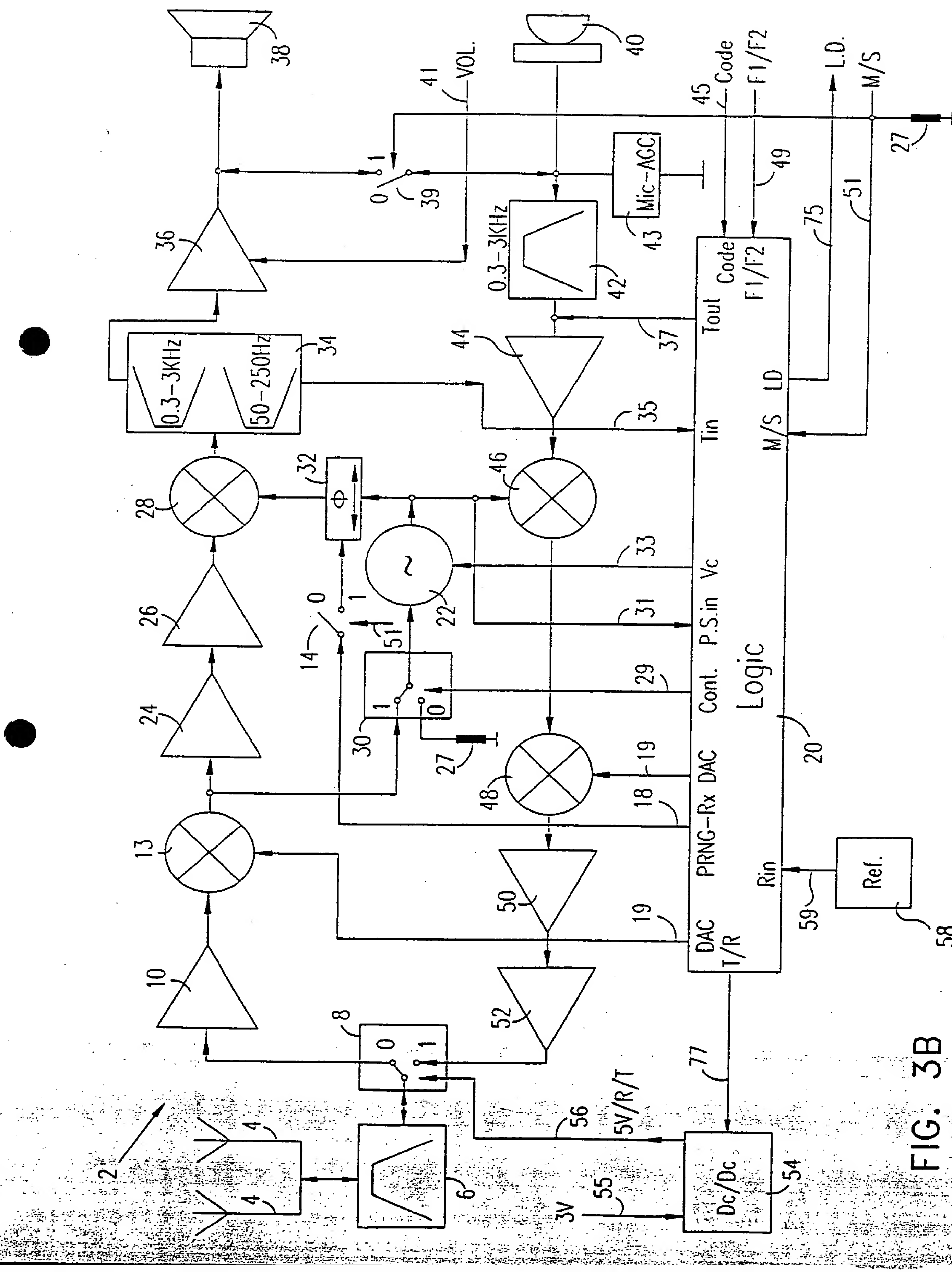

VOL.
Code
F1/F2
L.D.
M/S
Mic-AGC
0.3-3KHz
0.3-3KHz
50-250Hz
Tout
Tin
LD
M/S
Vc
P.S.in
Cont.
Logic
DAC
PRNG-Rx
Rin
DAC
T/R
Ref.
5V/R/T
3V
Dc/Dc

FIG. 3B

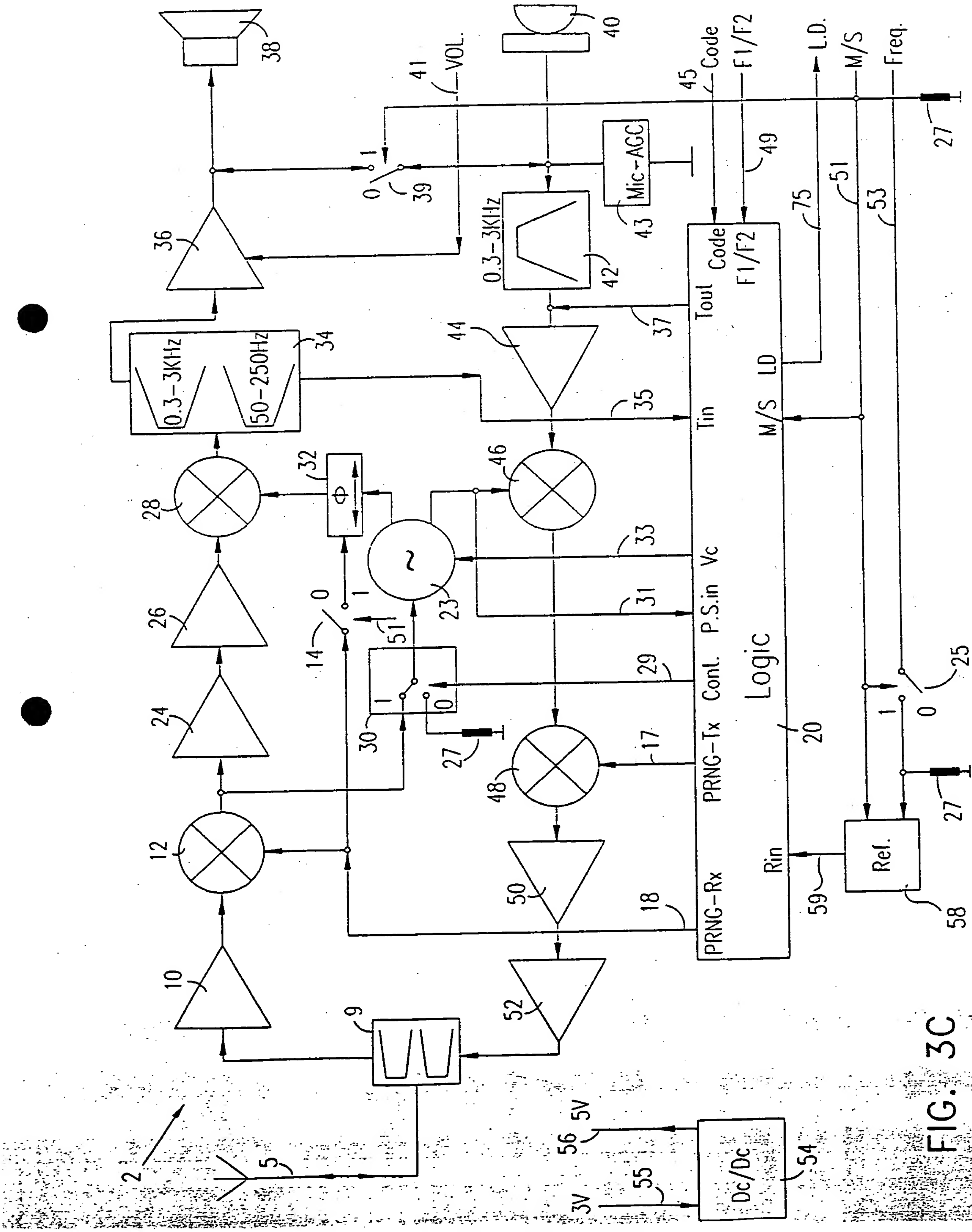

VOL.
Code
F1/F2
L.D.
M/S
Freq.
Mic-AGC
0.3-3KHz
0.3-3KHz
50-250Hz
Code
F1/F2
Tout
LD
Tin
M/S
Vc
P.S.in
Cont.
PRNG-Tx
PRNG-Rx
Logic
Rin
Ref.
Dc/Dc
5V
3V

FIG. 3C

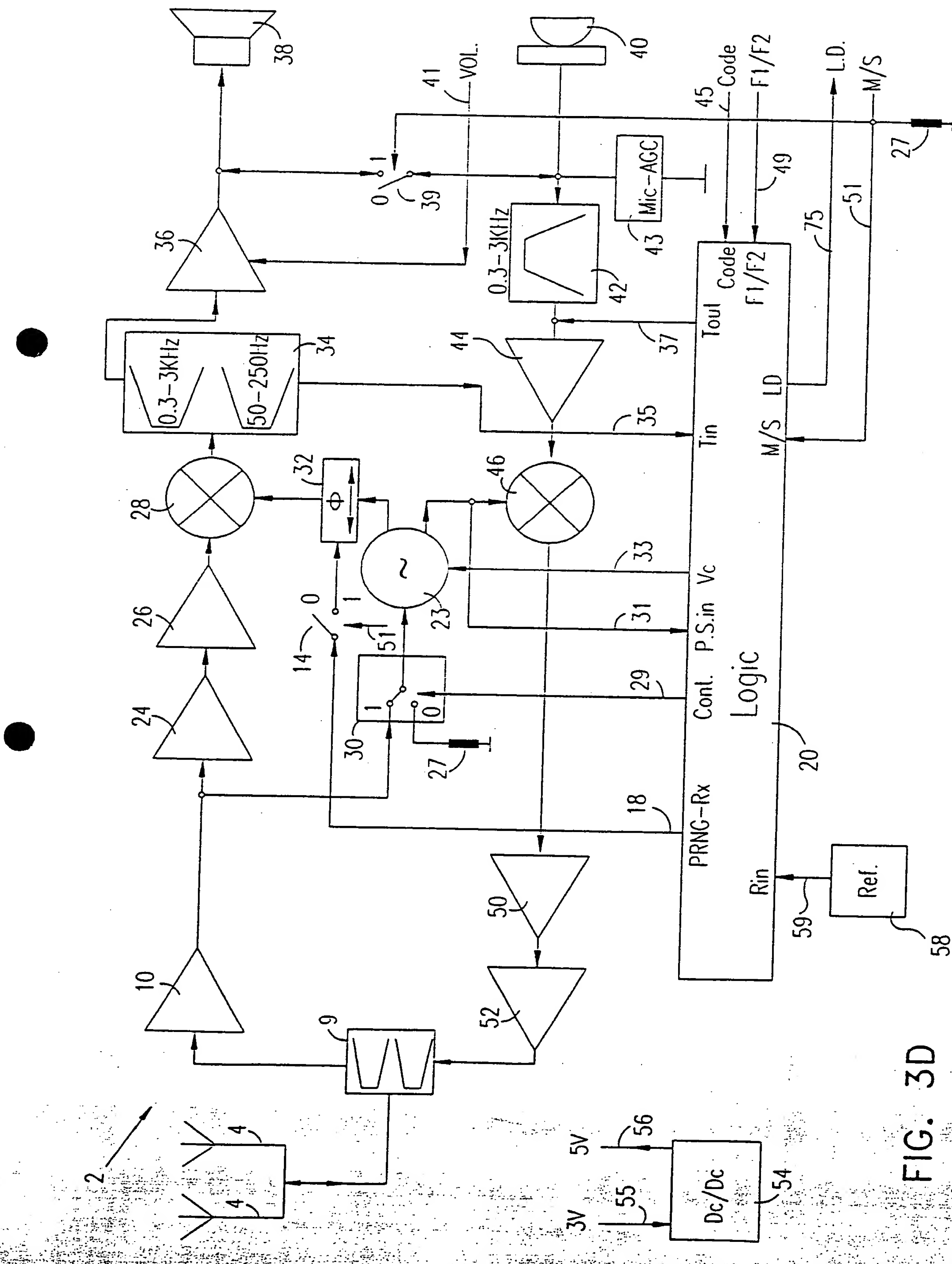

VOL.
Code
F1/F2
L.D.
M/S
Mic-AGC
0.3-3KHz
0.3-3KHz
50-250Hz
Code
F1/F2
Toul
LD
Tin
M/S
Vc
P.S.in
Cont.
Logic
PRNG-Rx
Rin
Ref.
Dc/Dc
5V
3V

FIG. 3D

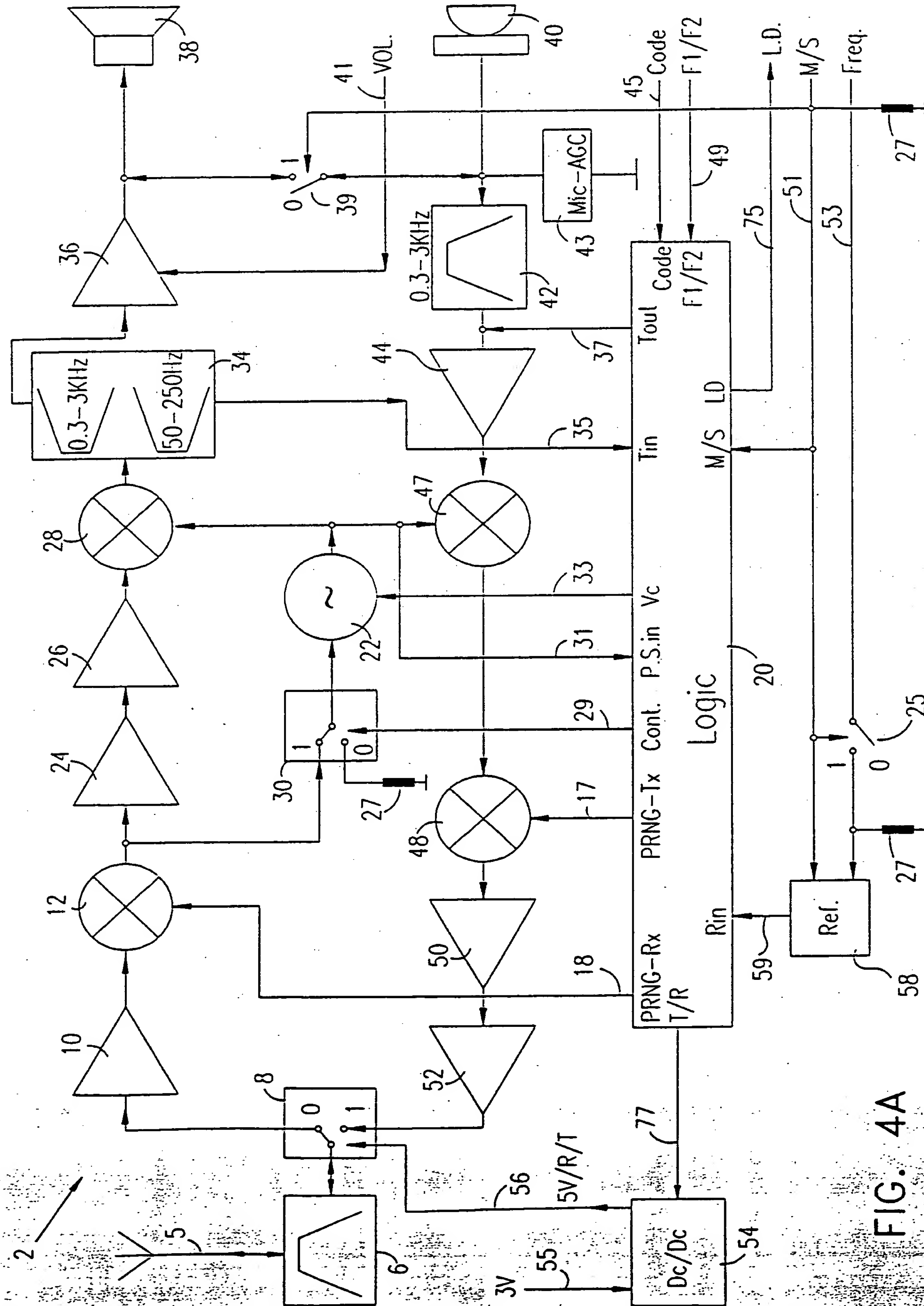

VOL.
Code
F1/F2
L.D.
M/S
Freq.
Mic-AGC
0.3-3KHz
50-250Hz
Tout
Tin
Vc
P.S.in
Cont.
PRNG-Tx
PRNG-Rx
T/R
Logic
LD
Rin
Ref.
Dc/Dc
3V
5V/R/T

FIG. 4A

DORON SHALEV

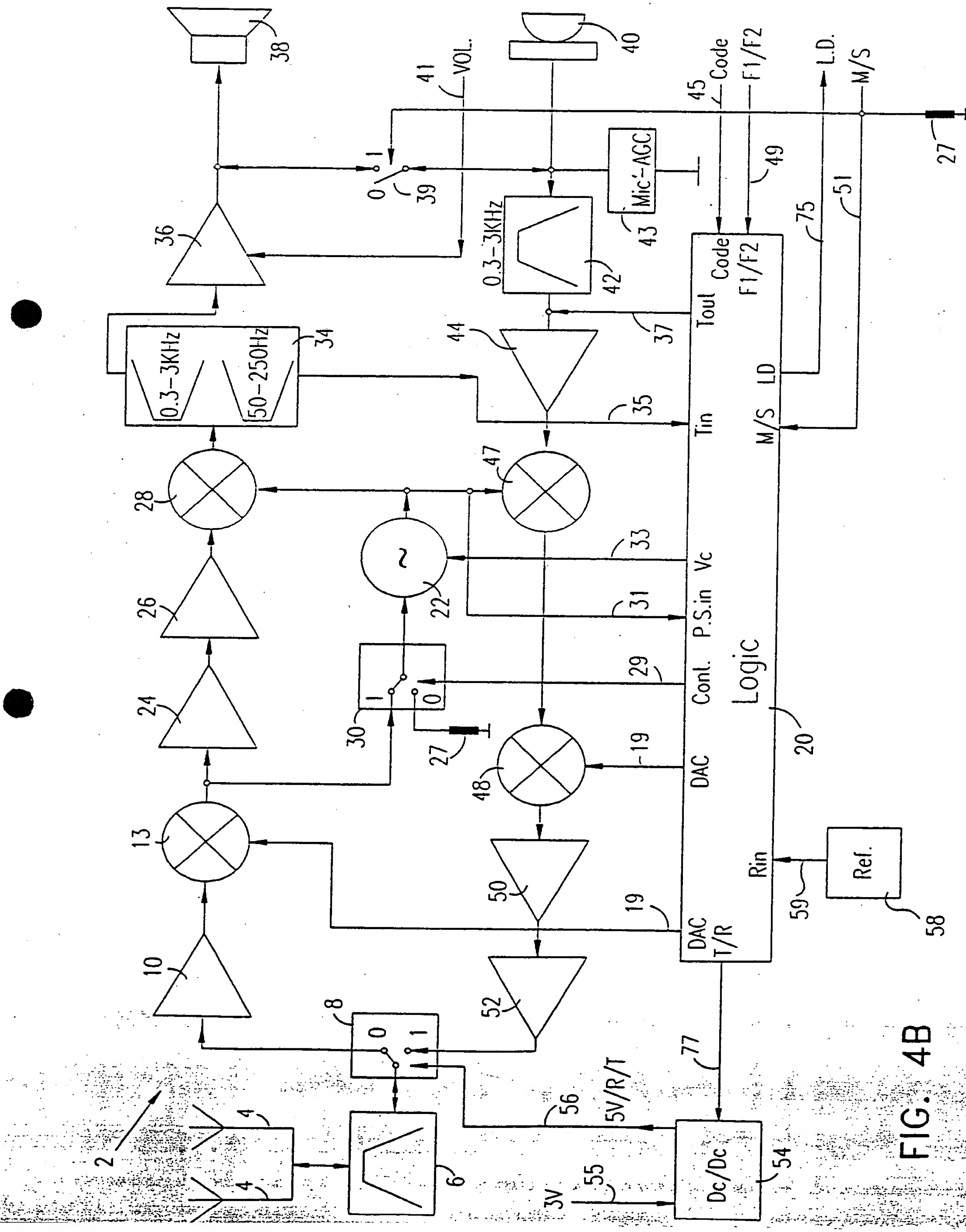
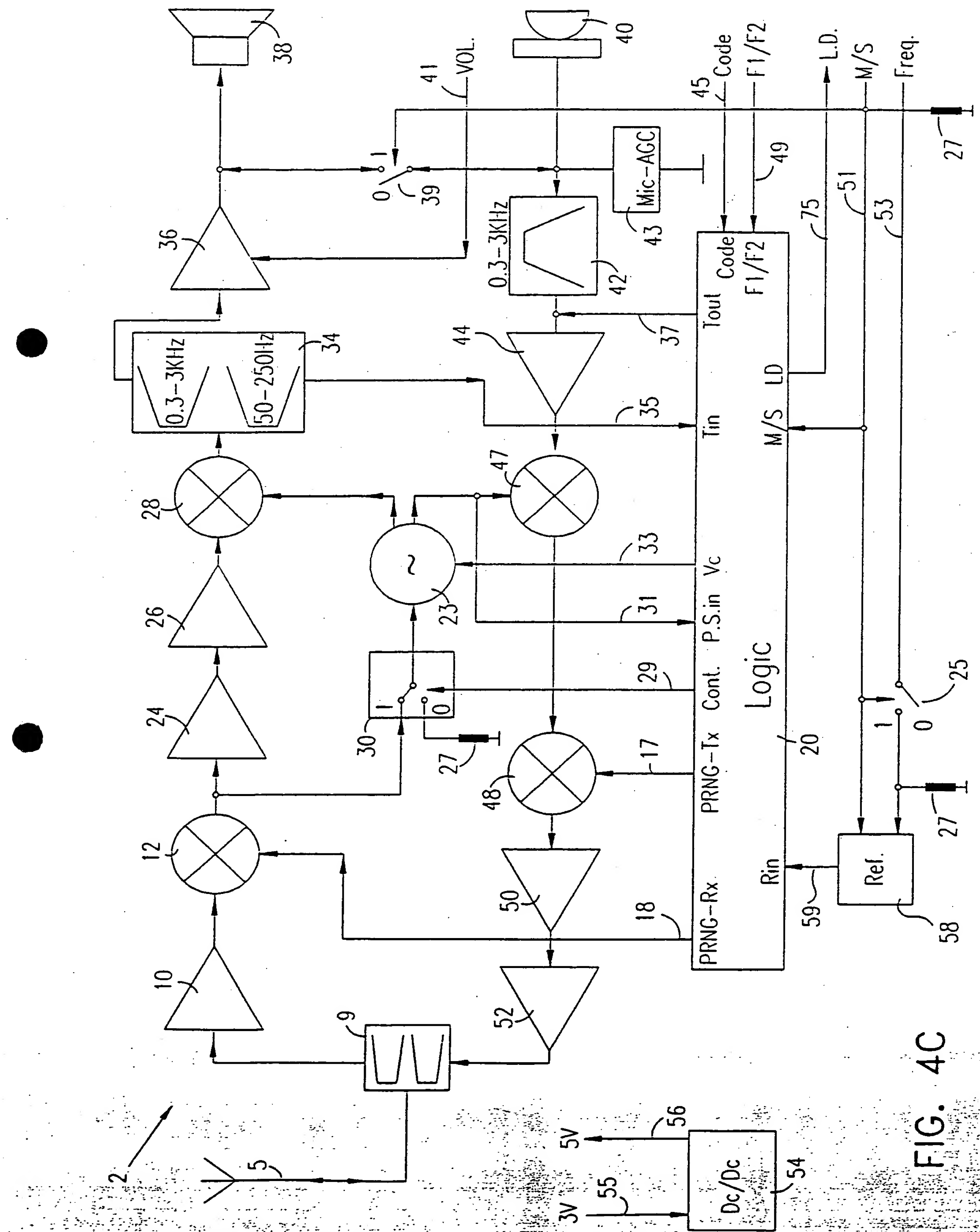


FIG. 4B



0.3-3KHz
50-250Hz
Mic-AGC
VOL.
Code
F1/F2
L.D.
M/S
Freq.
Tout
Tin
LD
Vc
P.S.in
Cont.
PRNG-Tx
PRNG-Rx
Rin
Logic
Ref.
Dc/Dc
3V
5V

FIG. 4C

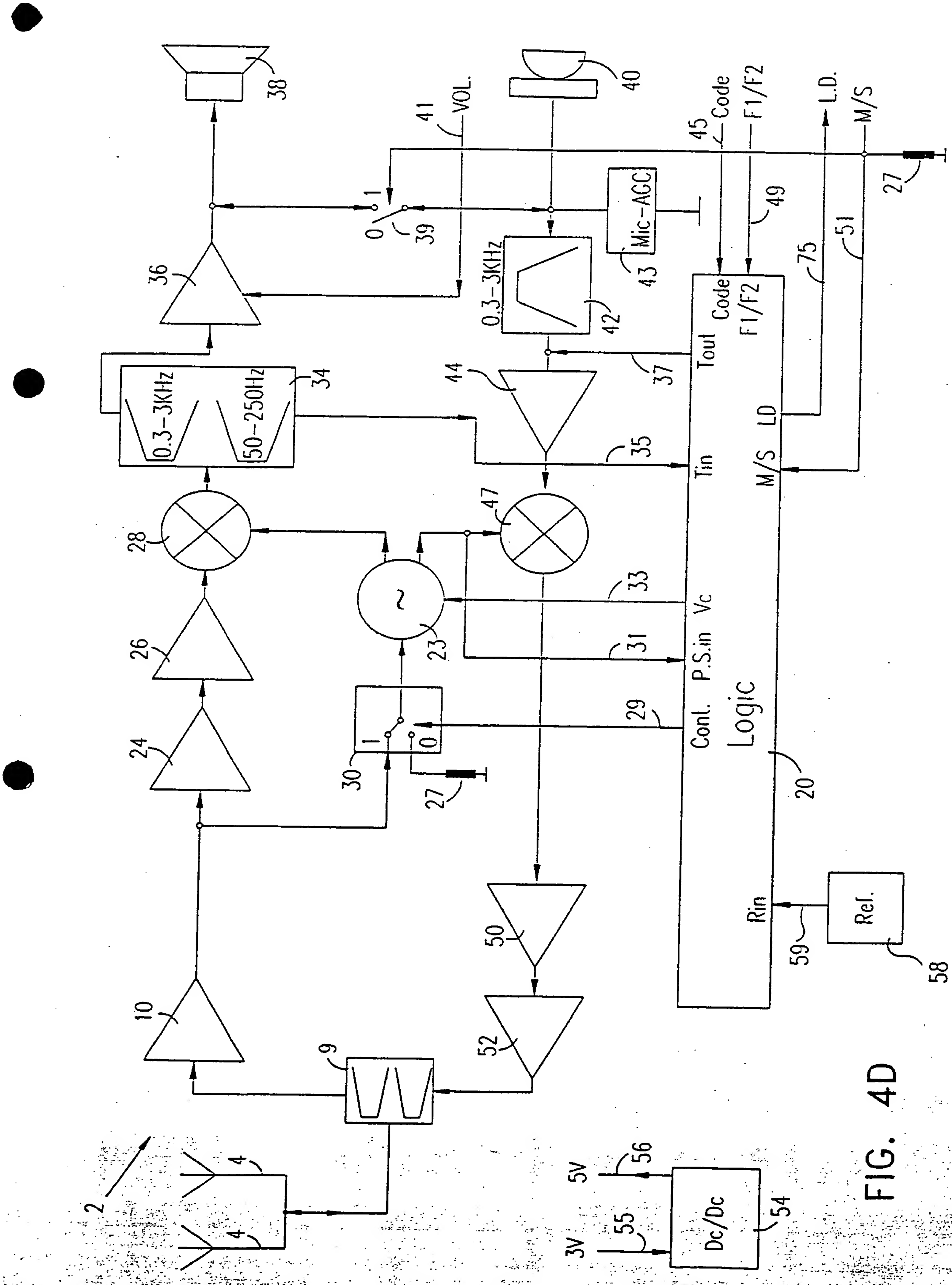

VOL.
Code
F1/F2
L.D.
M/S
Mic-AGC
0.3-3KHz
50-250Hz
Tout
Tin
LD
Vc
P.S.in
Cont.
Logic
Rin
Ref.
Dc/Dc
5V
3V

FIG. 4D

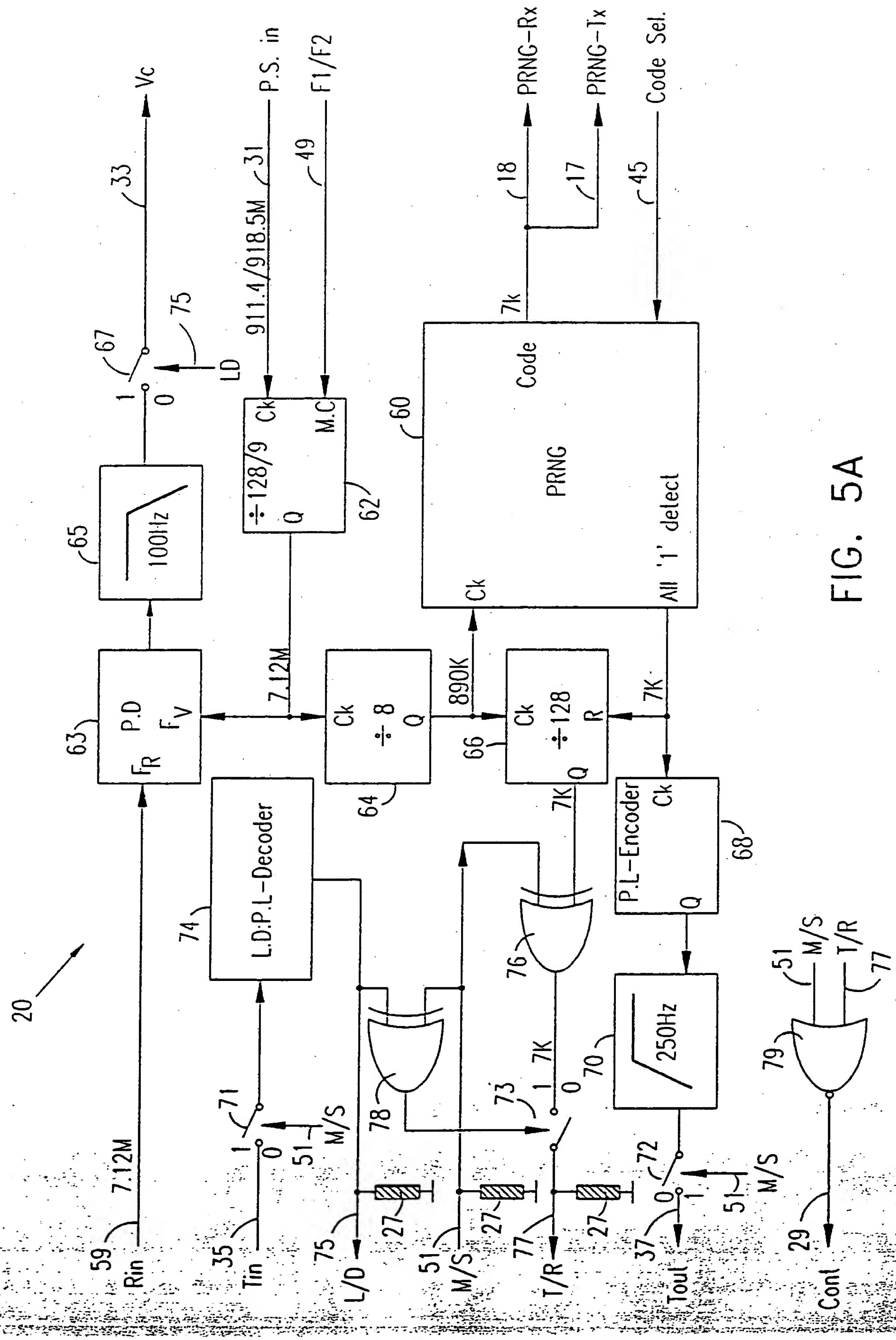

Vc
P.S. in
F1/F2
PRNG-Rx
PRNG-Tx
Code Sel.
911.4/918.5M
LD
÷128/9
Ck
M.C
Q
PRNG
Code
Ck
All '1' detect
100Hz
P.D
F_R
F_V
7.12M
÷ 8
890K
÷128
R
7K
L.D:P.L-Decoder
P.L-Encoder
250Hz
M/S
T/R
Rin
Tin
L/D
Tout
Cont

FIG. 5A

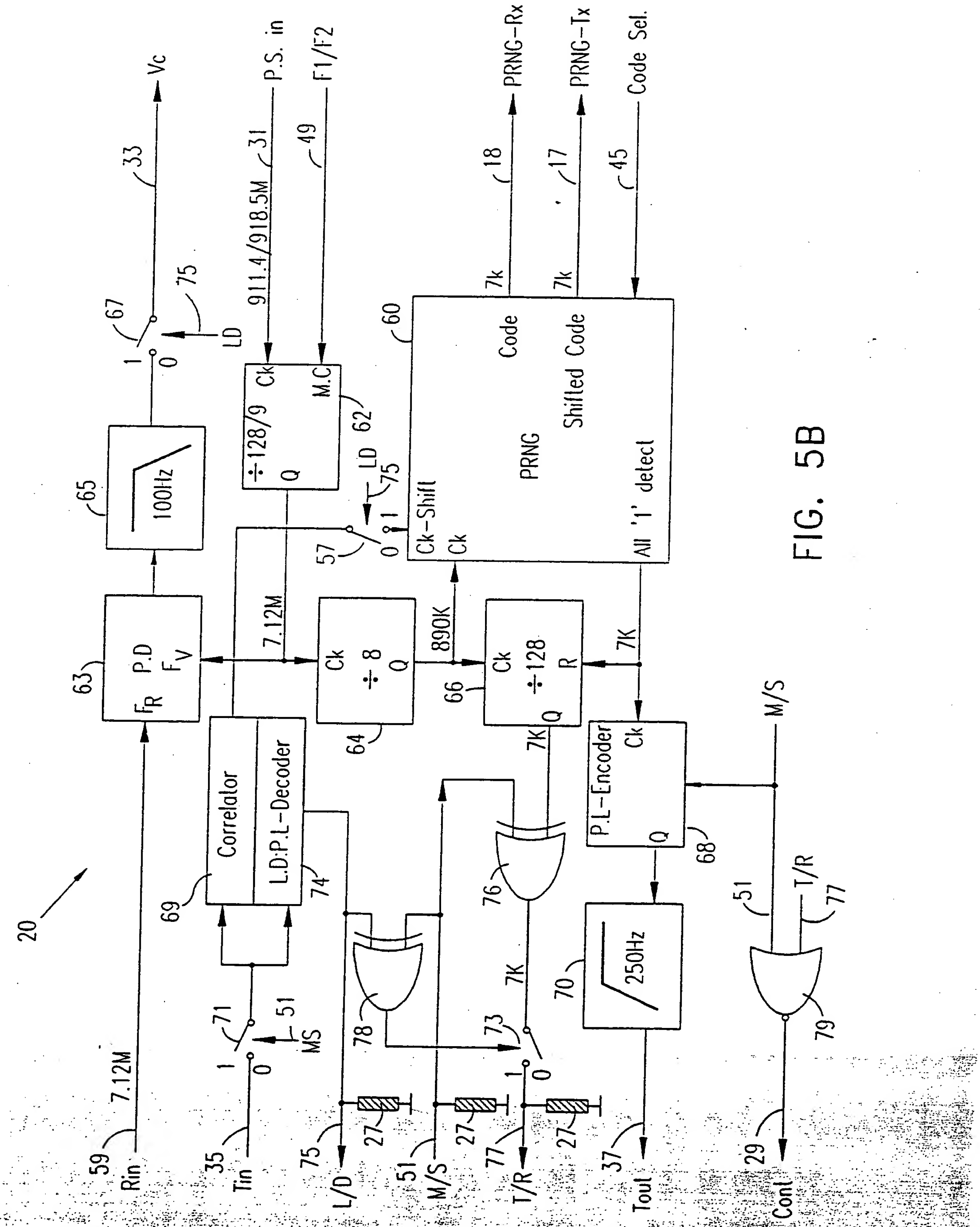

20
Rin
59
7.12M
Tin
35
L/D
75
M/S
51
T/R
77
Tout
37
Cont
29
27
P.D
F_R
F_V
63
100Hz
65
Vc
33
67
LD
75
911.4/918.5M
P.S. in
31
F1/F2
49
÷128/9
Ck
M.C
Q
62
7.12M
Correlator
L.D:P.L-Decoder
69
74
Ck
÷ 8
Q
64
890K
Ck
÷128
R
Q
66
7K
57
Ck-Shift
Ck
PRNG
Code
Shifted Code
All '1' detect
60
7k
PRNG-Rx
18
PRNG-Tx
17
Code Sel.
45
P.L-Encoder
68
250Hz
70
76
78
71
MS
73
79

FIG. 5B

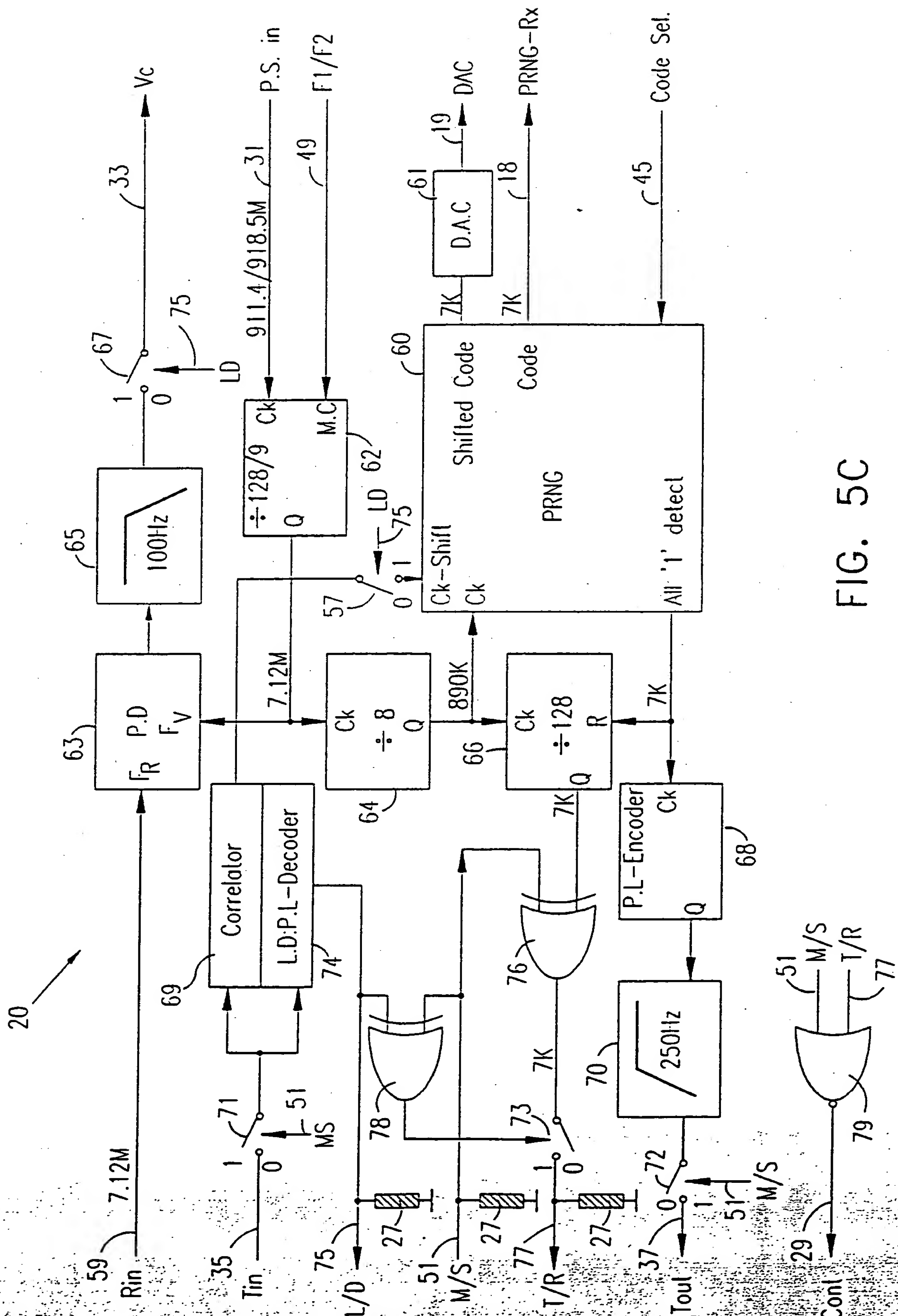

Vc
P.S. in
F1/F2
DAC
PRNG-Rx
Code Sel.
911.4/918.5M
D.A.C
7K
Shifted Code
Code
PRNG
All '1' detect
Ck-Shift
Ck
÷128/9
Ck
M.C
Q
LD
100Hz
P.D
F_R
F_V
7.12M
÷8
890K
÷128
R
Correlator
L.D:P.L-Decoder
P.L-Encoder
250Hz
M/S
T/R
MS
Rin
Tin
L/D
Tout
Cont
20
33
31
49
19
61
18
45
67
75
60
62
57
65
63
64
66
68
69
74
76
78
71
51
73
72
70
79
77
59
35
27
37
29

FIG. 5C

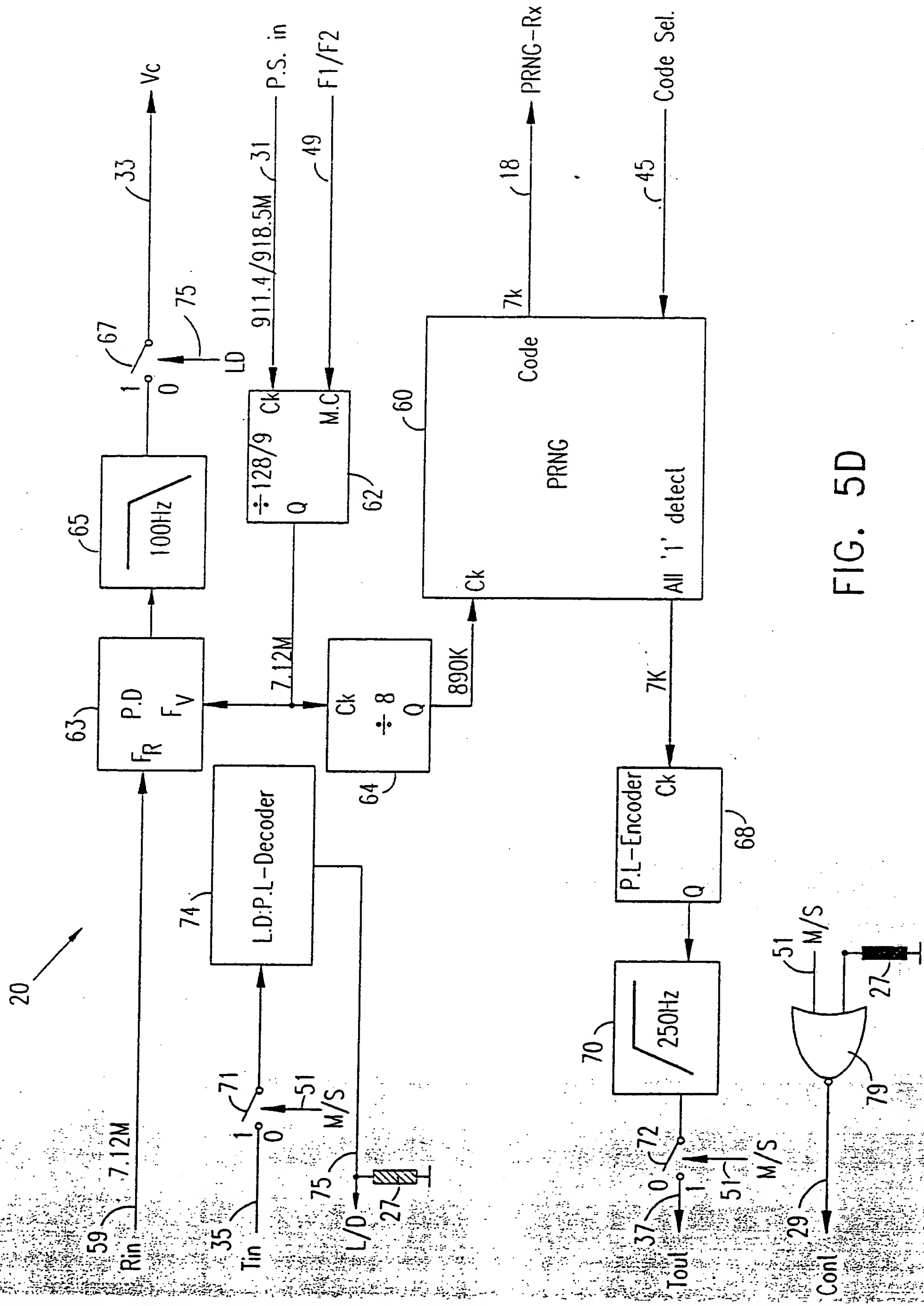

20
Rin
59
7.12M
P.D
F_R
F_V
63
100Hz
65
1
0
67
LD
75
33
Vc
7.12M
Ck
÷ 8
Q
64
890K
÷ 128/9
Ck
M.C
Q
62
911.4/918.5M
31
P.S. in
49
F1/F2
PRNG
60
Code
Ck
All '1' detect
7k
18
PRNG-Rx
45
Code Sel.
7K
P.L-Encoder
Ck
Q
68
250Hz
70
L.D:P.L-Decoder
74
Tin
35
71
51
M/S
L/D
75
27
Tout
37
72
Conl
29
79

FIG. 5D

FIG. 6A

M/S	L.D	T/R	Cont.	Vc
1	0	1,0,1...	0,0,0...	Enabled
0	0	0,0,0...	1,1,1...	Enabled
0	1	0,1,0...	1,0,1...	Disabled

Legend

M/S=0->Slave
M/S=1->Master
T/R=0->Rx
T/R=1->Tx

FIG. 6B

M/S	L.D	Cont.	Vc
1	0	0	Enabled
0	0	1	Enabled
0	1	1	Disabled

Legend

M/S=0->Slave
M/S=1->Master

FIG. 7

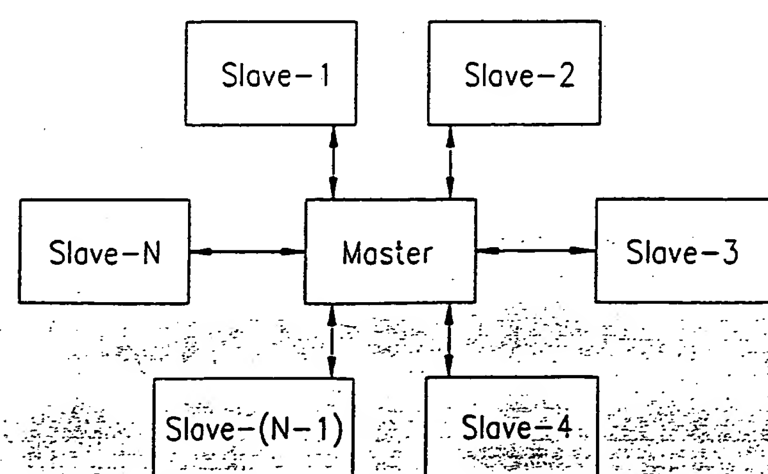


FIG. 8A

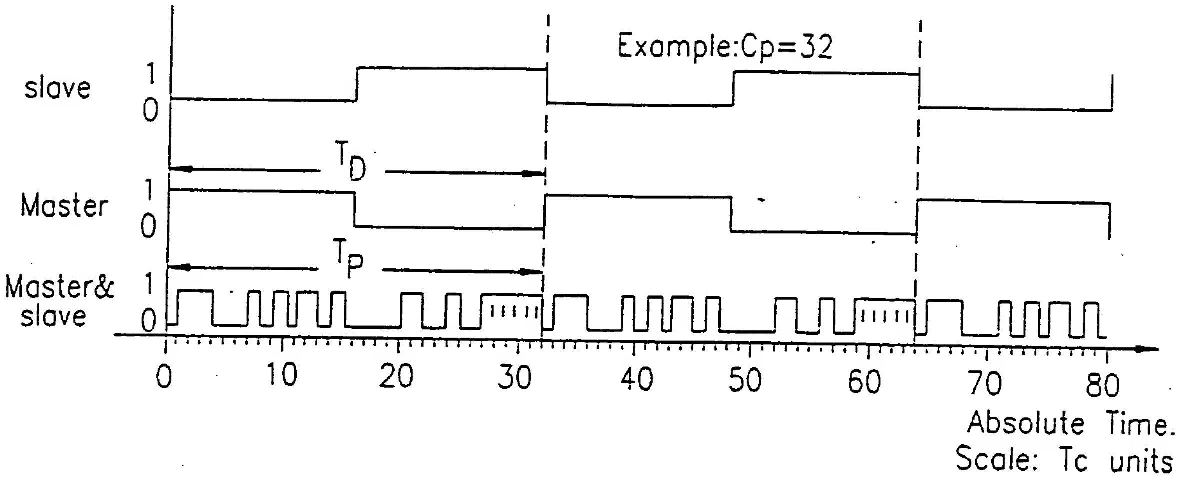


FIG. 8B

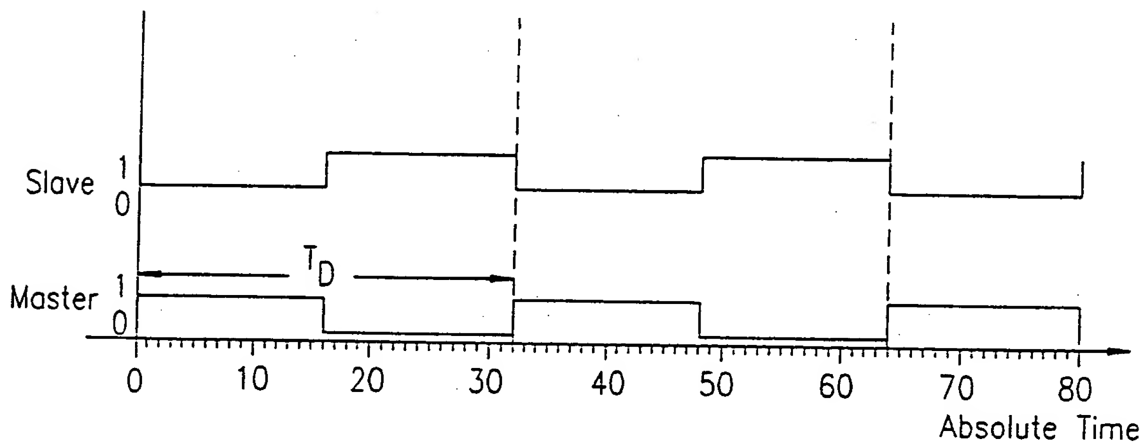


FIG. 8C

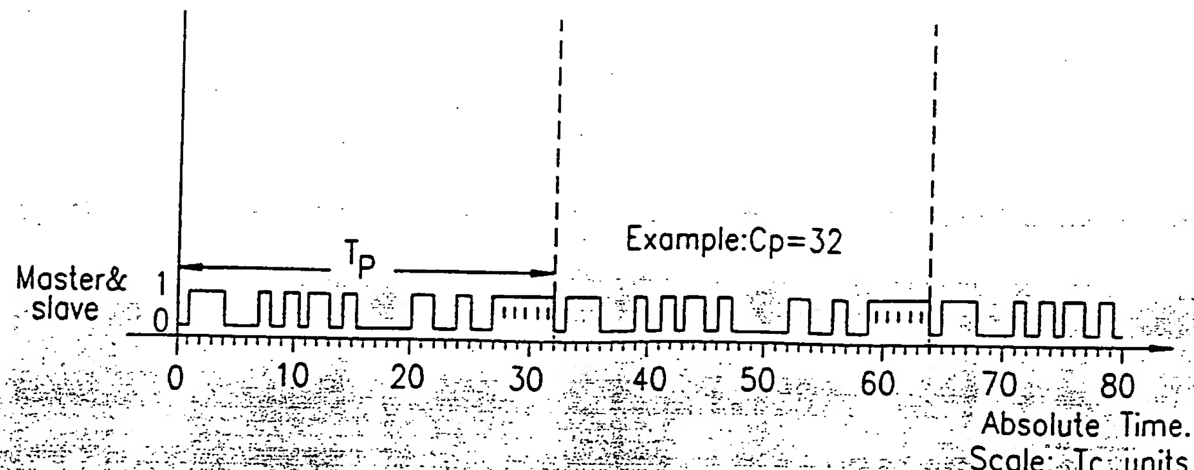


FIG. 9A

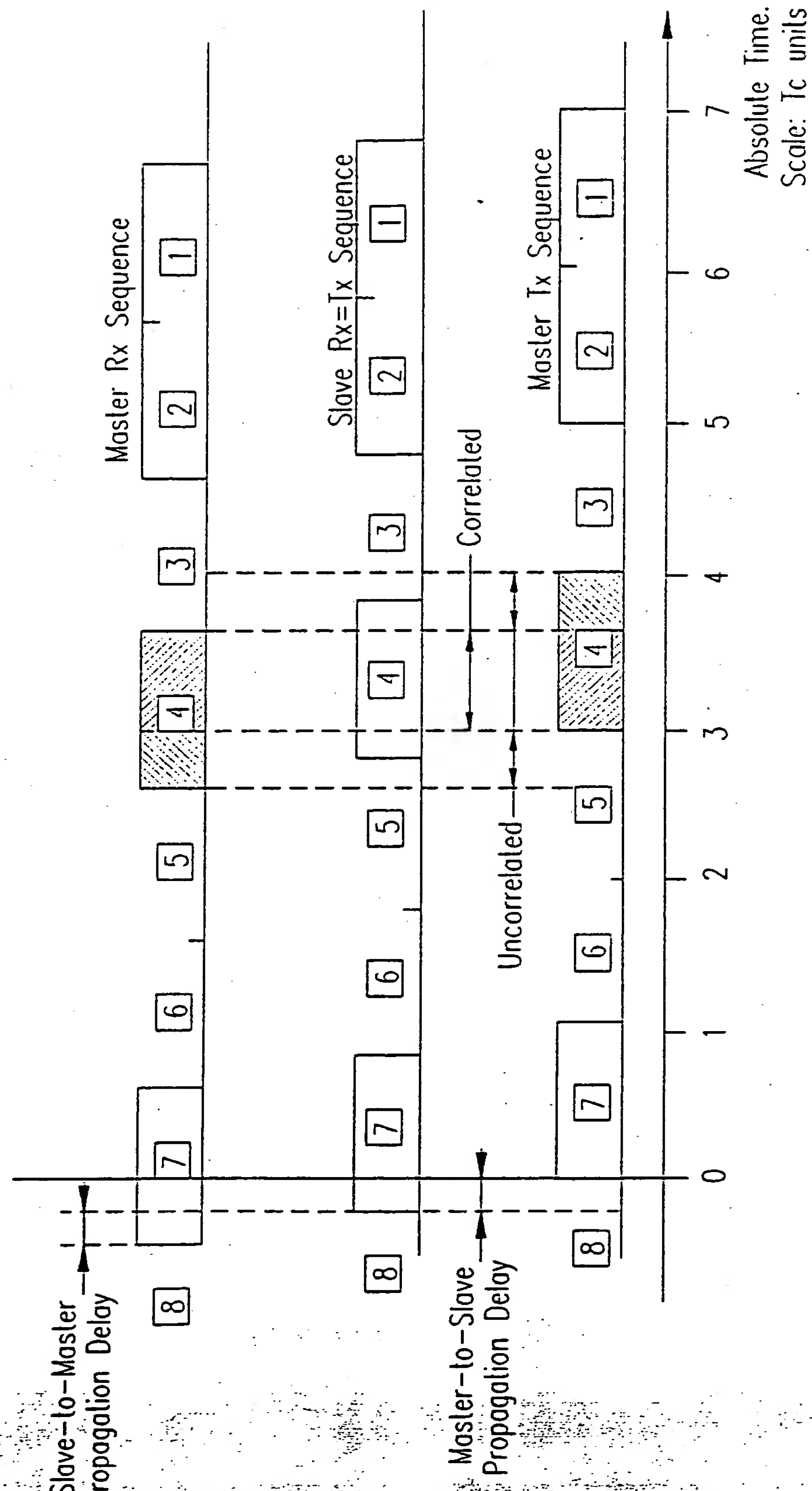

Master Rx Sequence
Slave Rx=Tx Sequence
Master Tx Sequence
Correlated
Uncorrelated
Slave-to-Master Propagation Delay
Master-to-Slave Propagation Delay
Absolute Time.
Scale: Tc units
0
1
2
3
4
5
6
7

FIG. 9B

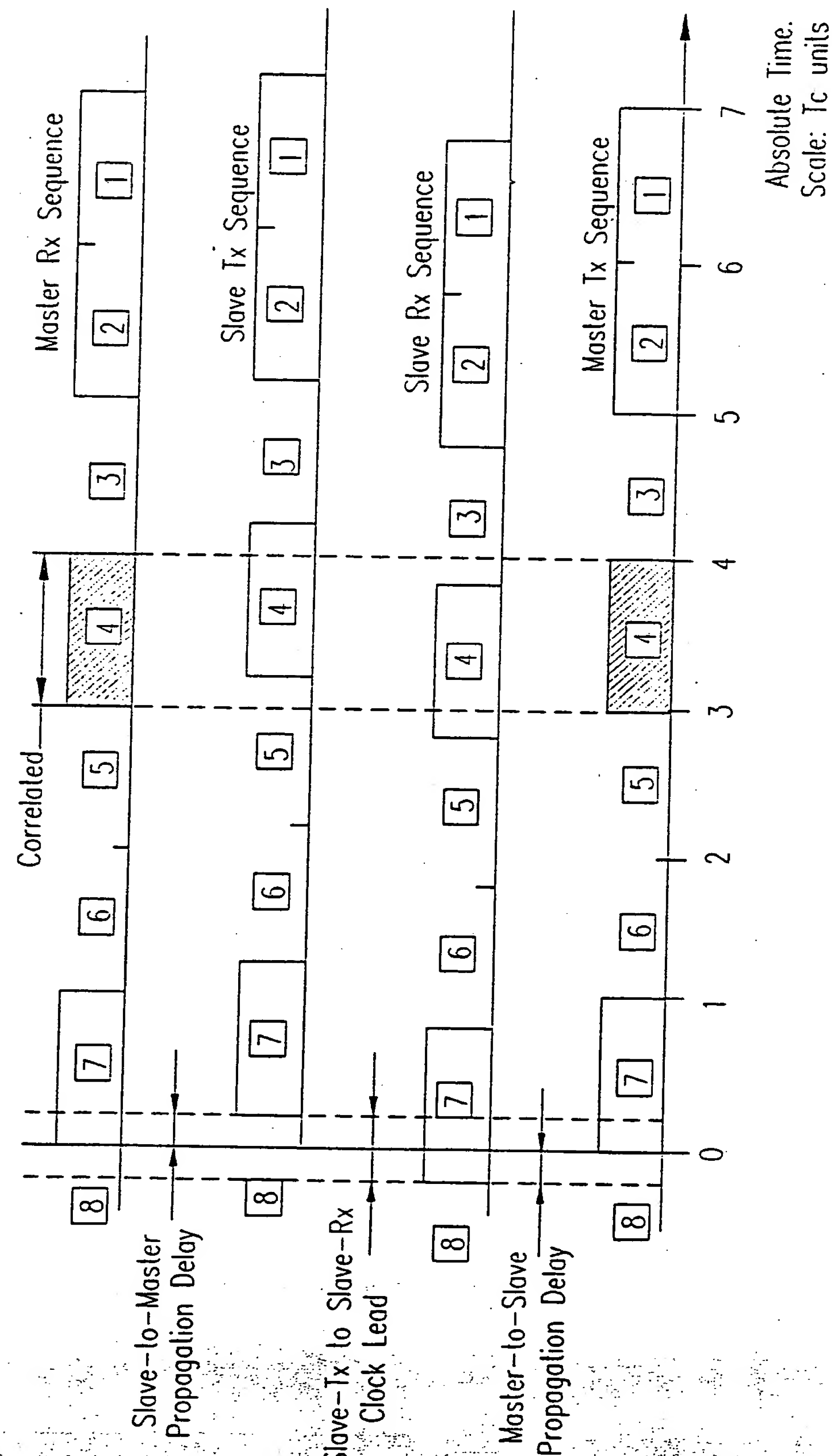
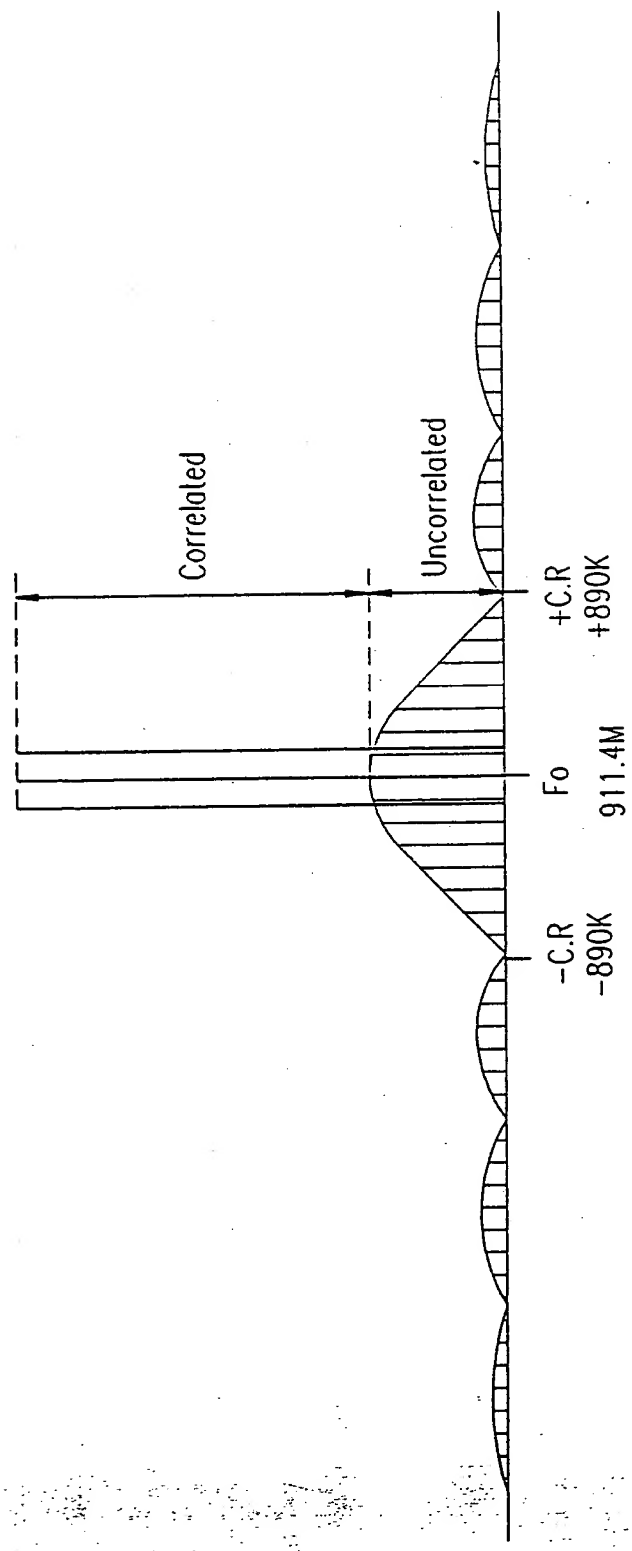

Master Rx Sequence
Slave Tx Sequence
Slave Rx Sequence
Master Tx Sequence
Correlated
Slave-to-Master Propagation Delay
Slave-Tx to Slave-Rx Clock Lead
Master-to-Slave Propagation Delay
Absolute Time.
Scale: Tc units
0
1
2
3
4
5
6
7

FIG. 10



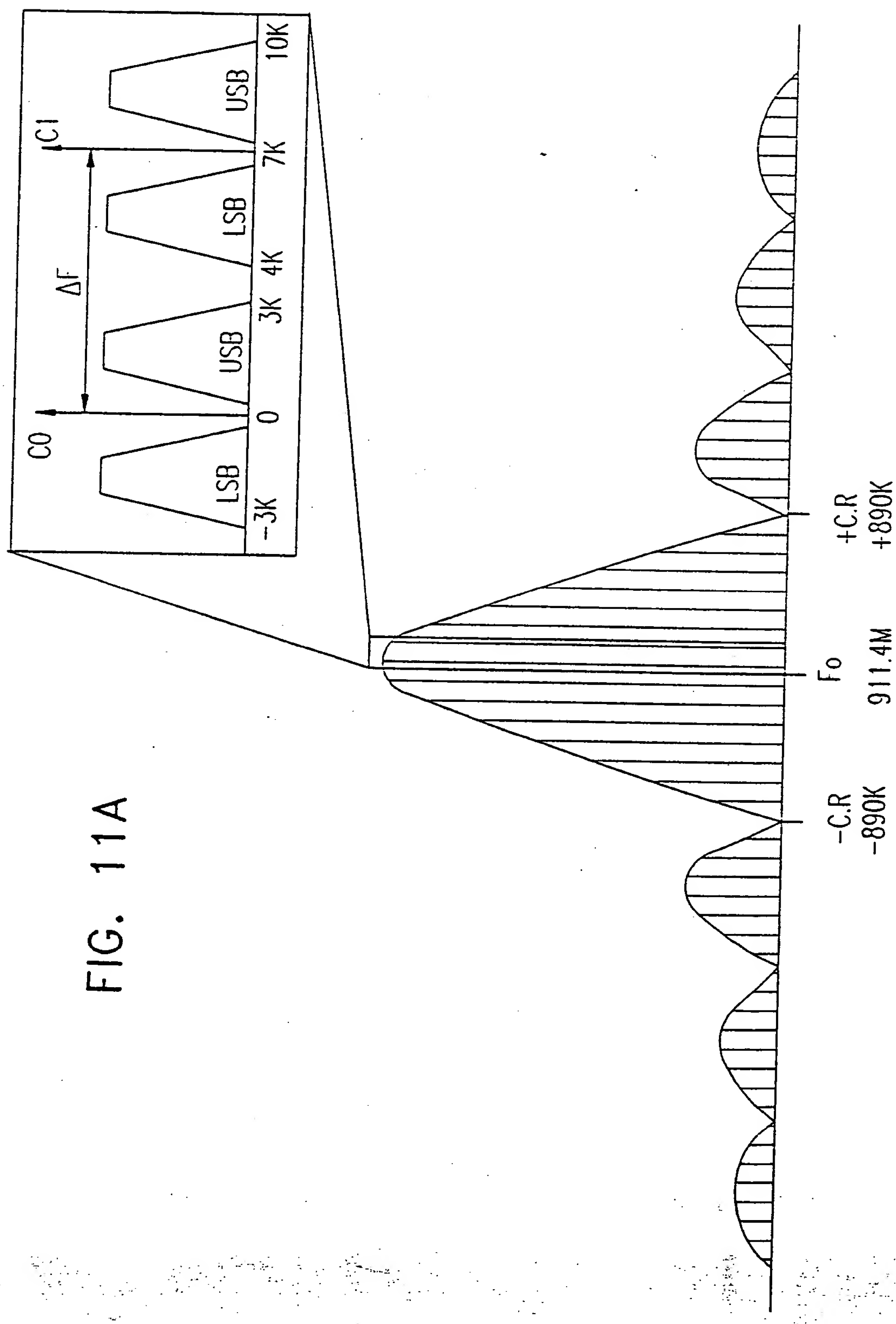
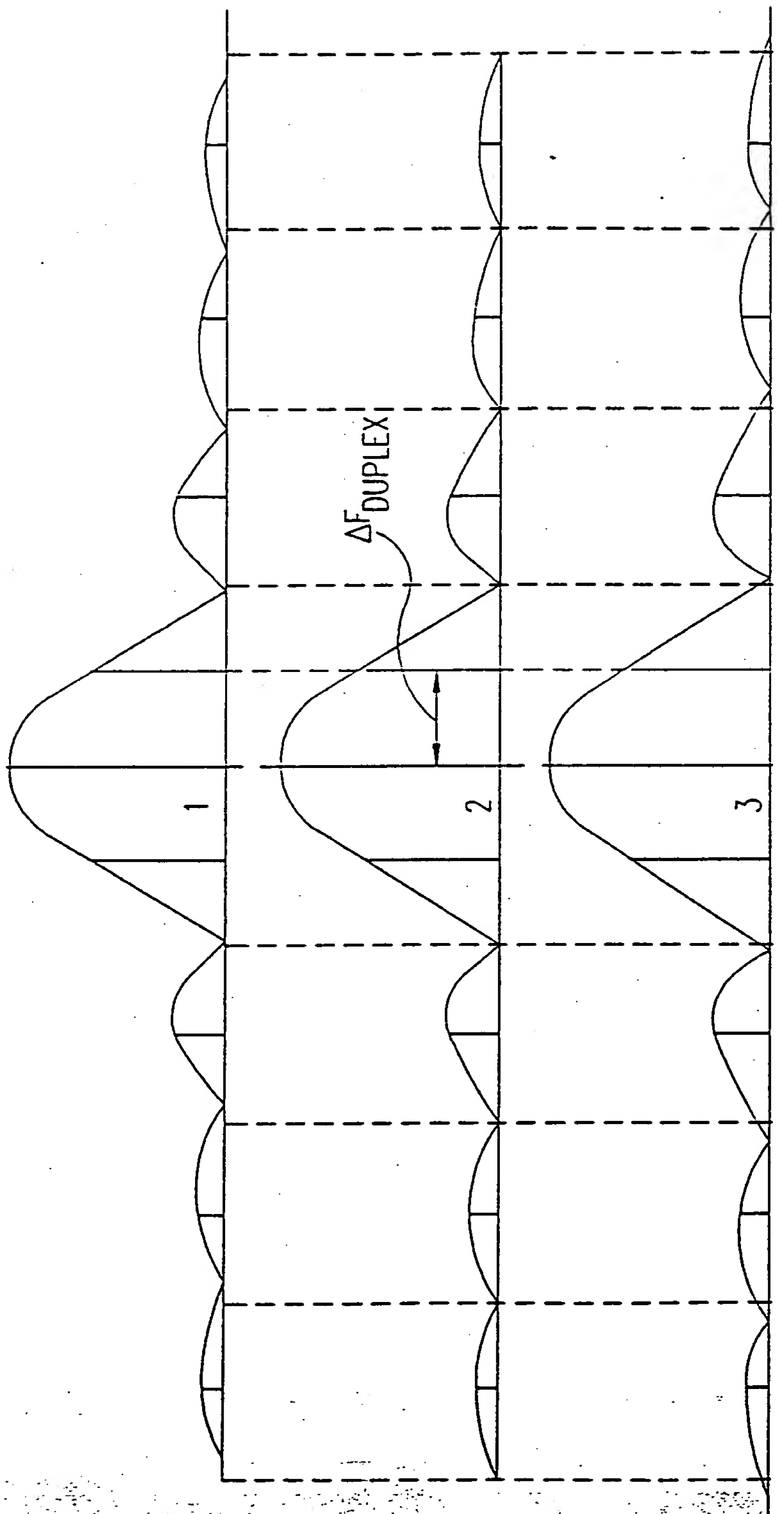


FIG. 11A

FIG. 11B



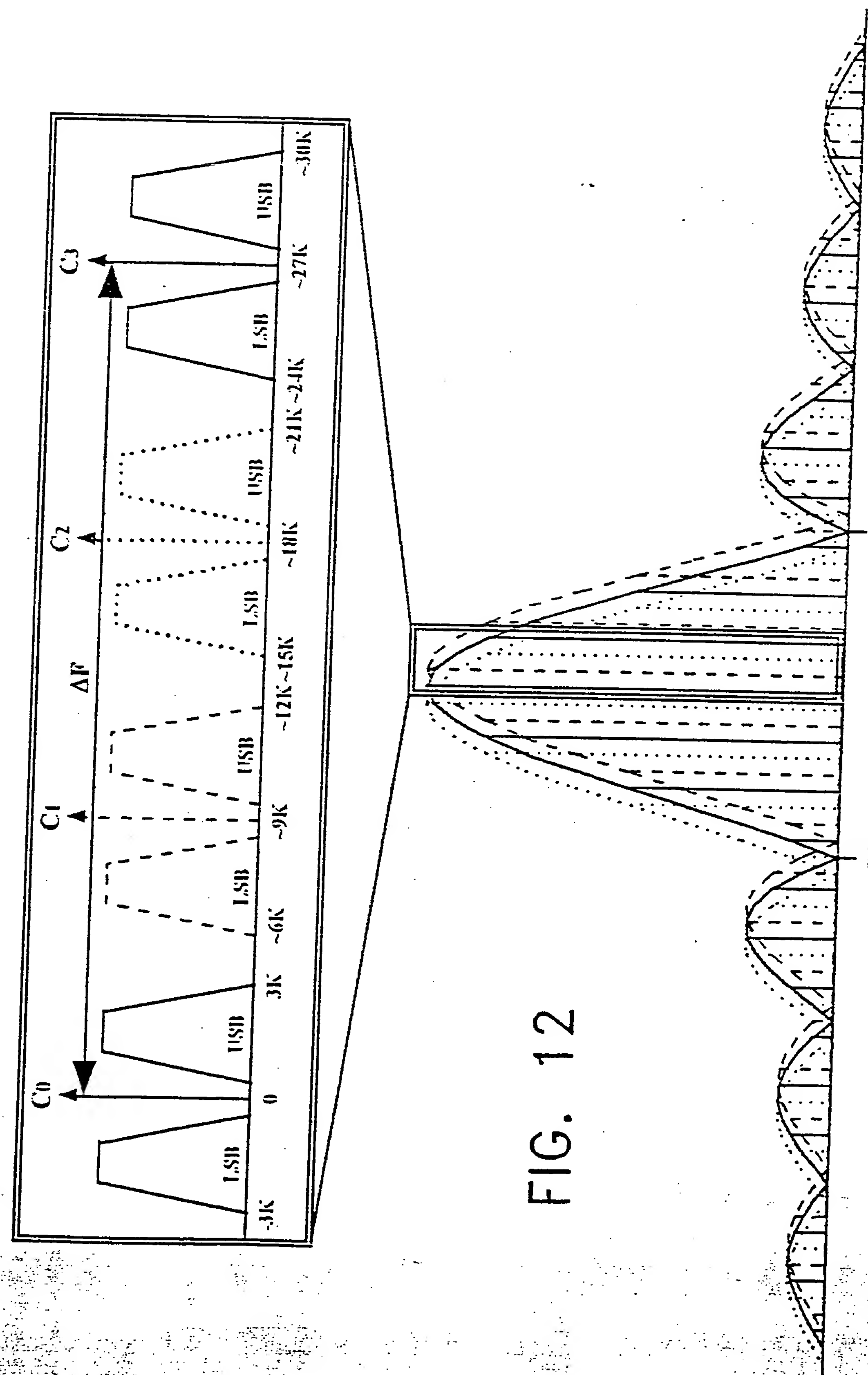

C0
C1
C2
C3
ΔF
LSB
USB
LSB
USB
LSB
USB
LSB
USB
-3K
0
3K
~6K
~9K
~12K~15K
~18K
~21K~24K
~27K
~30K
-C.R
-890K
F0
911.4M
+C.R
+890K

FIG. 12

FIG. 13A

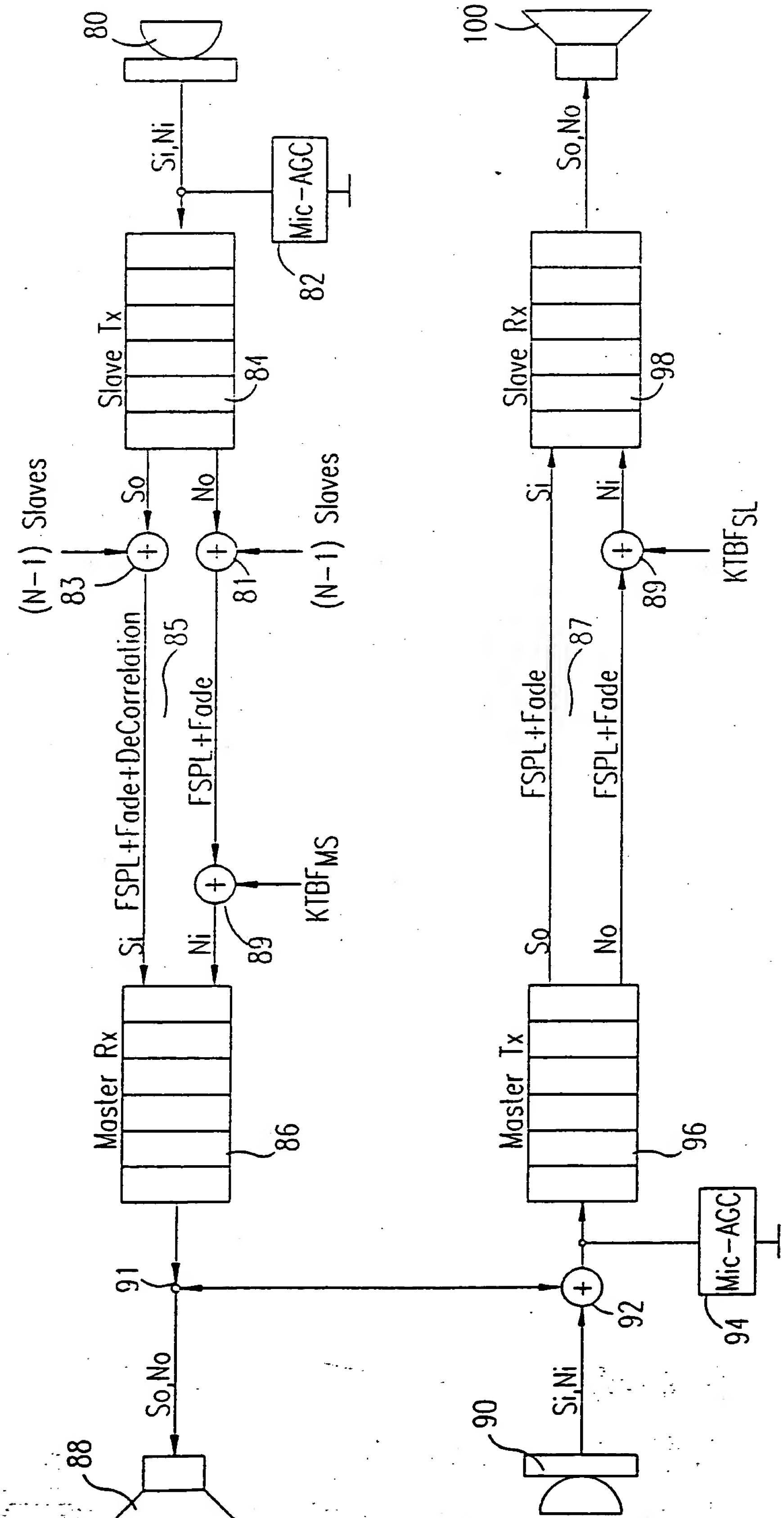

80
Si,Ni
Mic-AGC
82
Slave Tx
84
So
No
(N-1) Slaves
83
81
(N-1) Slaves
FSPL+Fade+DeCorrelation
85
FSPL+Fade
KTBF$_{MS}$
89
Si
Ni
Master Rx
86
91
So,No
88
100
So,No
Slave Rx
98
Si
Ni
89
KTBF$_{SL}$
FSPL+Fade
87
FSPL+Fade
So
No
Master Tx
96
Mic-AGC
94
92
Si,Ni
90

FIG. 13B

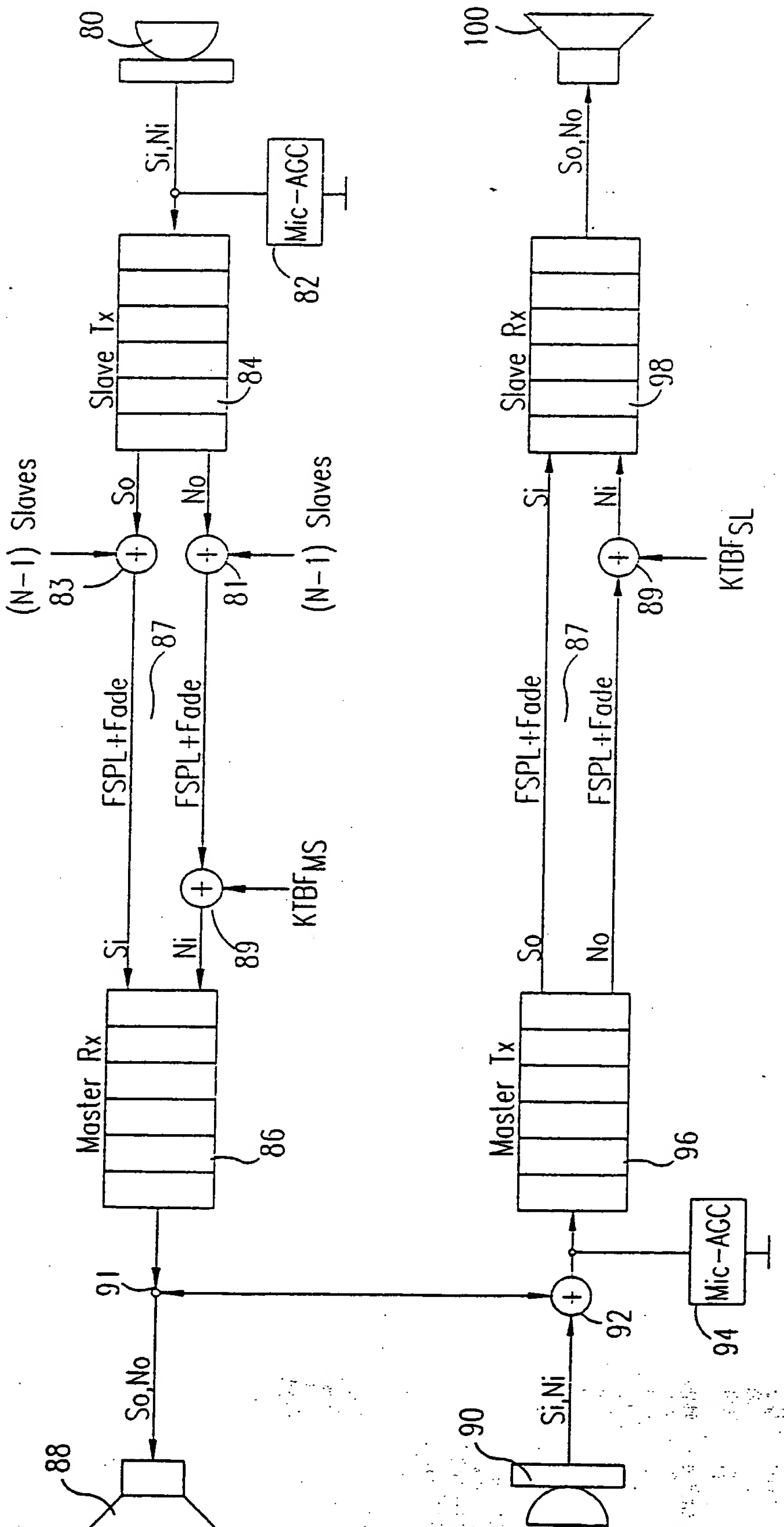

80
Si,Ni
Mic-AGC
82
Slave Tx
84
So
No
(N-1) Slaves
83
81
(N-1) Slaves
87
FSPL+Fade
FSPL+Fade
KTBF$_{MS}$
89
Si
Ni
Master Rx
86
91
So,No
88
100
So,No
Slave Rx
98
Si
Ni
KTBF$_{SL}$
89
87
FSPL+Fade
FSPL+Fade
So
No
Master Tx
96
Mic-AGC
94
92
Si,Ni
90

FIG. 14